MultiVOIP® GSM

SIP-to-Cellular Gateway

Model: MVPGSM-2

User Guide

MultiTech® Systems

User Guide

S000450C

Wireless MultiVOIP GSM Unit (Model: MVPGSM-2)

This publication may not be reproduced, in whole or in part, without prior expressed written permission from Multi-Tech Systems, Inc. All rights reserved.

Copyright © 2010, by Multi-Tech Systems, Inc.

Multi-Tech Systems, Inc. makes no representations or warranty with respect to the contents hereof and specifically disclaims any implied warranties of merchantability or fitness for any particular purpose. Furthermore, Multi-Tech Systems, Inc. reserves the right to revise this publication and to make changes from time to time in the content hereof without obligation of Multi-Tech Systems, Inc. to notify any person or organization of such revisions or changes. Check Multi-Tech's Web site for current versions of our product documentation.

Record of Revisions

Revision	Date	Description
A	11/04/08	Initial release.
B	07/07/09	Added new features.
C	01/22/10	New release version 15.12.2A. Added portal and power information.

Patents

This Product is covered by one or more of the following U.S. Patent Numbers: ***6219708***, ***6151333, 5757801, 5682386***, ***5.301.274; 5.309.562***; ***5.355.365; 5.355.653; 5.452.289; 5.453.986***. Other Patents Pending.

Trademark

Registered trademarks of Multi-Tech Systems, Inc. are MultiVOIP GSM, Multi-Tech, and the Multi-Tech logo. Windows is a registered trademark of Microsoft.

World Headquarters

Multi-Tech Systems, Inc.
2205 Woodale Drive, Mounds View, Minnesota 55112
Phone: 763-785-3500 or 800-328-9717 Fax: 763-785-9874
www.multitech.com

Contacting Multi-Tech Support

Online Support Portal

https://support.multitech.com

In order to better serve our customers, manage support requests and shorten resolution times, we have created the online web portal allowing you to submit questions regarding Multi-Tech products directly to our technical support team. Get answers to your most complex questions, ranging from implementation, troubleshooting, product configuration, firmware upgrades and much more.

To create an account and submit a Support Case on the Portal, visit support.multitech.com.

Knowledge Base and Support Services

www.multitech.com/en_US/SUPPORT

The Knowledge Base provides immediate answers to your questions and gives you access to support resolutions for all Multi-Tech products. Visit our support area on the website for other support services.

Technical Support

Country	By Email	By Phone
Europe, Middle East, Africa:	support@multitech.co.uk	(44) 118 959 7774
U.S., Canada, all others:	support@multitech.com	(800) 972-2439 or (763) 717-5863

Warranty

Warranty information can be found at: http://www.multitech.com/en_US/COMPANY/Policies/warranty/

CONTENTS

CHAPTER 1 – DESCRIPTION AND SPECIFICATIONS 4
Specifications 7
Unpacking Your MultiVOIP GSM 8
Safety Warnings 9
CHAPTER 2 – GETTING STARTED 10
Installing SIM Cards 10
Modem Numbering 10
Cabling Procedure 11
GSM Instructions 11
Mounting Instructions 13
CHAPTER 3 – MULTIVOIP SOFTWARE 14
Installing the MultiVOIP GSM Software 14
Basic Setup 17
CHAPTER 4 – CONFIGURING YOUR MULTIVOIP GSM 31
Software Categories Covered in This Chapter 31
How to Navigate Through the Software 32
Web Browser Interface 32
CHAPTER 5 – PHONE BOOK CONFIGURATION 69
Sample Configuration: IP to Cellular Call Flow 70
Sample Configuration: Cellular to IP Call Flow 73
Phone Book Descriptions 84
CHAPTER 6 – USING THE SOFTWARE 92
Software Categories Covered in This Chapter 92
Statistics Section 94
MultiVOIP GSM Program Menu Items 106
FTP Server File Transfers (“Downloads”) 111
Web Browser Interface 115
SysLog Server Functions 118
APPENDIX A – PORTS & CABLE PIN-OUTS 119
APPENDIX B – REGULATORY INFORMATION 120
APPENDIX C – ADDITIONAL PHONE BOOK EXAMPLES 122
APPENDIX D – WASTE ELECTRICAL AND ELECTRONIC EQUIPMENT (WEEE) STATEMENT 124
APPENDIX E – C-ROHS HT/TS SUBSTANCE CONCENTRATION 125
INDEX 126

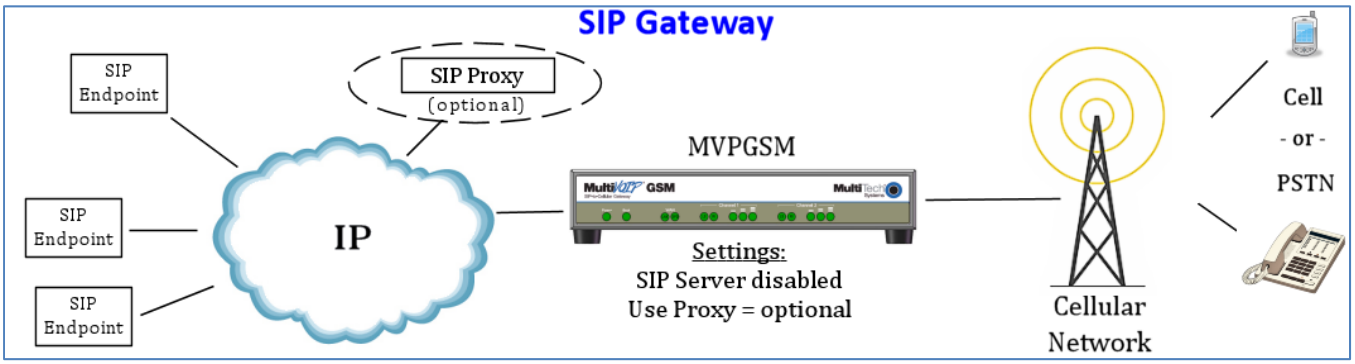
Chapter 1 – Description and Specifications

Introduction

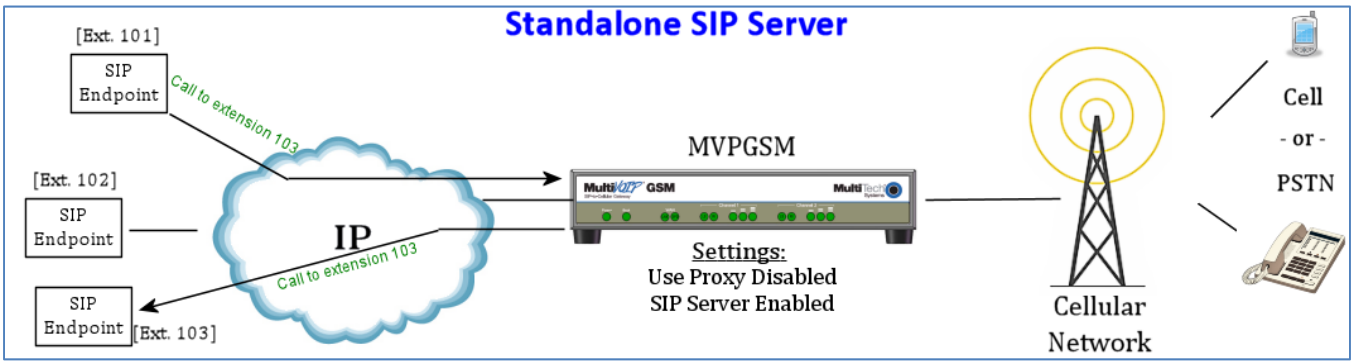
The MultiVOIP GSM provides wireless voice communications over the Internet or an Intranet. By integrating wireless connectivity into your existing data network, you can realize substantial savings on inter-office long distance toll charges. The MVPGSM has "phone books," which are directories set up to simulate dialing and connecting as though the call was in the local area.

The MVPGSM can be used in several ways – four examples are graphically depicted below.

SIP Gateway Mode:
The MVPGSM functions as a SIP gateway similar to the other SIP endpoints shown in the drawing below.

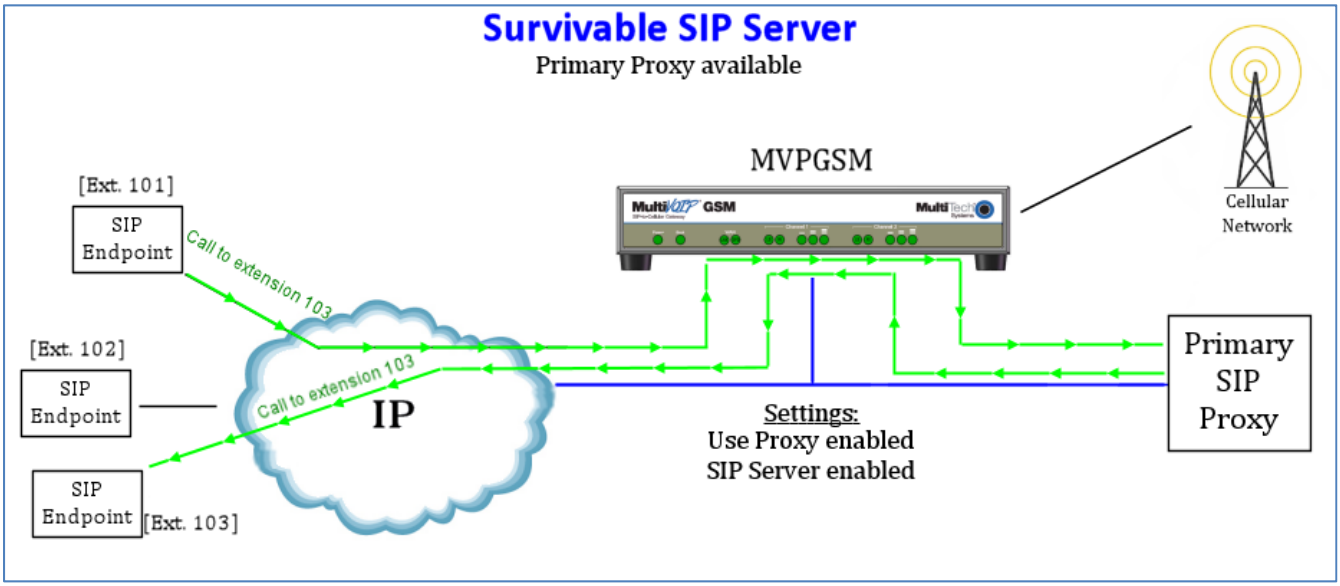


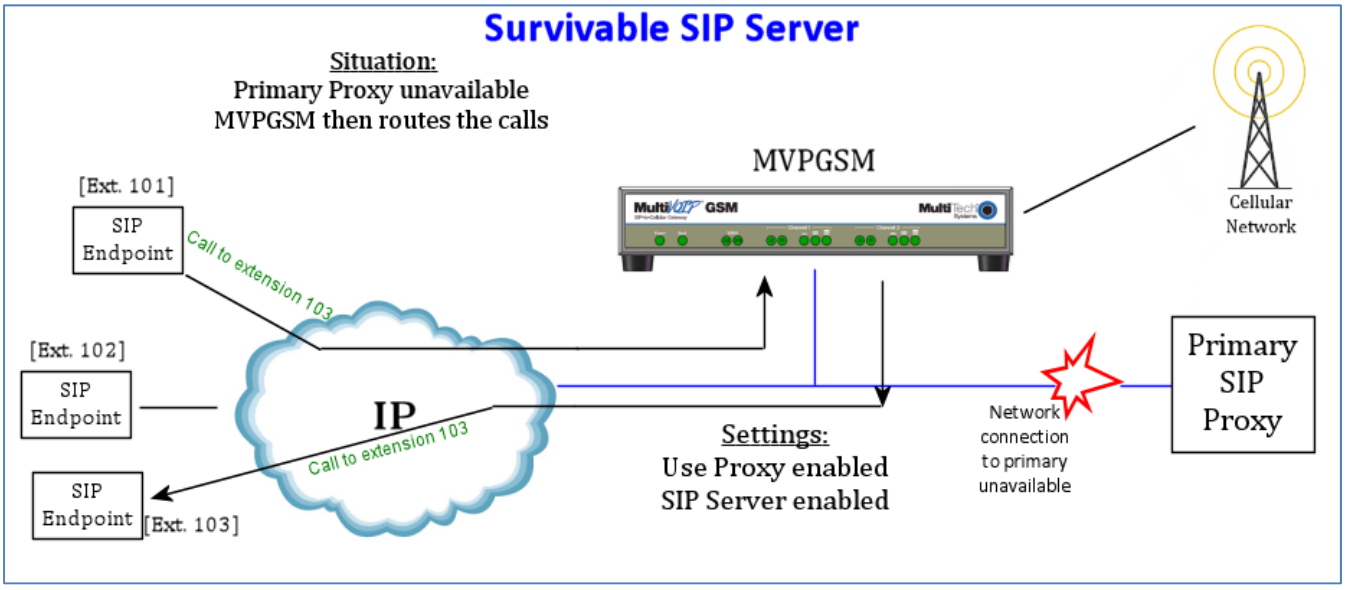
As a Standalone SIP Server:
The MVPGSM functions as the SIP server for all of the SIP endpoints. Enable the SIP Server on the MVPGSM and configure a predefined endpoint matching each SIP endpoint or enable the "Allow Undefined Registrations" from any SIP endpoint.



As a Survivable SIP Server:

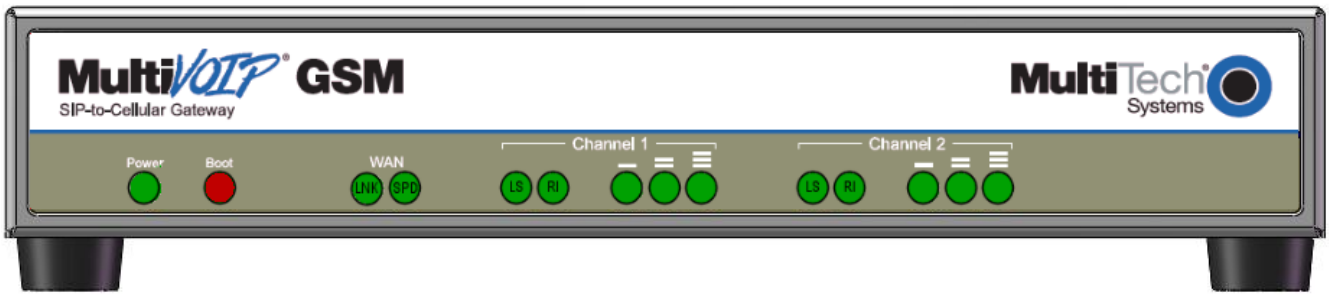
In this mode the MVPGSM functions as a Back to Back User Agent (B2BUA) and routes SIP registration and call control packets between the SIP endpoints and Primary proxy. The MVPGSM monitors the status of the primary proxy and, if the primary proxy is unavailable, the MVPGSM functions as a SIP server and takes over the routing of calls between SIP endpoints. When the network connection to the primary SIP proxy is restored, the MVPGSM resumes routing registration and call processing packets to the Primary SIP Proxy. SIP endpoints must be configured to use the MVPGSM as their proxy. On the MVPGSM, the SIP Server and Use Proxy options must be enabled. Predefined endpoints should be configured in the MVPGSM SIP Server so that the MVPGSM has the authentication information needed for re-registering SIP endpoints to the primary proxy.





Front Panel LEDs

The table below describes what the LED indicators represent.



Front Panel LED Definitions	
LED	**Description**
Power	Indicates presence of power
Boot	After power up, the Boot LED will be on while the MultiVOIP GSM is booting. It lights whenever the MultiVOIP GSM is booting, saving a configuration or receiving a firmware upgrade.
WAN	**LNK.** Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While the link is up, this LED will flash off to indicate activity. **SPD.** Speed indicator LED: this is lit when link speed is 100MB/s and off when it is 10MB/s.
Channel-Operation LEDs (one set for each channel)	

LS	**Line Status.**
	• **Continuous "on" state** indicates that the channel is connected.
	• **Flashing states:**
	  • 600 ms on / 600 ms off – searching for network, no SIM card or authentication in progress
	  • 75 ms on / 3000 ms off – IDLE mode; registered on network but no call in progress
	  • 75 ms on / 75 ms off / 75 ms on / 3000 ms off – GPRS PDP contexts activated
	• **Off state**. Channel is off (not ready) or in sleep mode.

RI	**Ring Indicator**. This LED will light when an incoming Ring Signal is detected.
Signal Strength	
Bars (0-3)	There are three LEDs for each channel. If no LEDs are lit, there is not enough signal strength to maintain the wireless connection reliably. If at least one LED is lit, there is enough signal to maintain connection with Low signal strength. If the left two LEDs are lit, a Medium signal strength is present. If all three LEDs are lit, a High signal strength is present.

Computer Requirements

The computer on which the MVPGSM configuration program is installed must meet these requirements:

- must be IBM-compatible PC with MS Windows operating system;
- must have an available COM port for connection to the MultiVOIP GSM.

However, this PC does not need to be connected to the MultiVOIP GSM permanently. It only needs to be connected when local configuration and monitoring are done. Nearly all configuration and monitoring functions can be done remotely via the IP network using a web browser.

Specifications

Operating Voltage/Current*	+9 to 12Vdc (9Vdc @ 750mA) (12Vdc @ 580mA)
Mains Frequencies	50 - 60 Hz
Power Consumption*	9 watts **Input 9 Vdc** **Typical** **Maximum** Current (Amps) 0.475 0.690 Watts 4.2 6.0 **Input 12 Vdc** **Typical** **Maximum** Current (A) 0.395 0.550 Watts 4.7 6.5
Mechanical Dimensions	1.71" H x (2.1" with feet attached) 9.792" W x 7.646" D ---------- 4.343 cm H x (5.334 cm with feet) 24.871 cm W x 19.42 cm D
Weight	4.75 lbs (2.14 kg)
Operational Temperature	-5° to +55° C (UL listed @ 40° C limited by power supply) @ 20-90% non-condensing relative humidity.
Certifications	**CE** EMC: FCC Parts 2/15/22/24, RSS 132/133, EN 301 489-1, EN 301 489-7, EN 301 511, EN 55022, EN 55024 Safety: UL 60950-1, cUL, IEC 60950-1, AS/NZS 60950:2000
Default IP Address	***192.168.3.143***
Warranty	2 years

***** Multi-Tech Systems, Inc. recommends that the customer incorporate a 10% buffer into their power source when determining product load.

Antenna Specifications

GSM/EGSM Antenna Requirements/Specifications

Frequency Range:	824 – 960 MHz / 1710 – 1990 MHz
Impedance:	50 Ohms
VSWR:	<2.0:1
Typical Radiated Gain:	3 dBi on azimuth plane
Radiation:	Omni
Polarization:	Vertical
Wave:	Half Wave Dipole

Antennas available from Multi-Tech Systems, Inc.

Description	Part Number
900/1800 MHz 1/2 Wave Antenna Mag Mount, 12.5", 1 Pack	ANF1-1MMHW
850/1900 MHz 1/2 Wave Antenna Mag Mount, 12.5", 1 Pack	ANF2-1MMHW

PTCRB Requirements Note

There cannot be any alteration to the authorized antenna system. The antenna system must be the same type with similar in-band and out-of-band radiation patterns and maintain the same specifications.

Antenna Note

The antenna gain, including cable loss, must not exceed 3.0 dBi at 1900 MHz / 1.4 dBi at 850 MHz for mobile operating configurations and 7.0 dBi at 1900 MHz / 1.4 dBi at 850 MHz for fixed mounted operations, as defined in 2.1091 and 1.1307 of the rules for satisfying RF exposure compliance.

The antenna(s) used for this transmitter must be installed to provide a separation distance of at least 20 cm from all persons and must not be collocated or operating in conjunction with any other antenna or transmitter.

Unpacking Your MultiVOIP GSM

When unpacking your MultiVOIP GSM, check to see that all of the items are included in the box.

MVPGSM content list:

- MVPGSM unit
- DE9 to RJ45 cable
- Power transformer
- Power cord
- Product CD
- Antenna

Safety Warnings

Lithium Battery Caution

A lithium battery on the voice/fax channel board provides backup power for the timekeeping capability. The battery has an estimated life expectancy of ten years. When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for replacement.

Warning: There is danger of explosion if the battery is incorrectly replaced.

General Safety

The MultiVOIP GSM is designed for, and intended to be used in fixed applications. “Fixed” means that the device is physically secured at one location and is not able to be easily moved to another location.

Caution: A separation distance of at least 20 cm (8 inches) must be maintained between the transmitter’s antenna and the body of the user or nearby persons. The MVPGSM is not designed for or intended to be used in portable applications within 20 cm. (8 inches) of the body of the user.

RF Interference Issues

It is important to follow any special regulations regarding the use of radio equipment due in particular to the possibility of radio frequency (RF) interference. Please follow the safety advice given below carefully.

- Switch OFF your MVPGSM in hospitals and any other place where medical equipment may be in use.
- Respect restrictions on the use of radio equipment in fuel depots, chemical plants or where blasting operations are in progress.
- There may be a hazard associated with the operation of your MVPGSM close to inadequately protected personal medical devices such as hearing aids and pacemakers. Consult the manufacturers of the medical device to determine if it is adequately protected.

Operation of your MVPGSM close to other electronic equipment may also cause interference if the equipment is inadequately protected. Observe any warning signs and manufacturers’ recommendations.

Chapter 2 – Getting Started

Introduction

The MultiVOIP GSM is equally usable as tabletop unit or mounted in a location with good reception. The initial setup is best performed before any mounting is done.

Installing SIM Cards

The MVPGSM requires at least one SIM card (Subscriber Identity Module) to operate on a GSM network. To install the SIM cards, do the following:

1. Turn the unit **off** and disconnect the power cord. Remove the screws from the back of the unit.

2. Electrostatic discharge (ESD) is the release of stored static electricity that can damage electrical circuitry or components. Static electricity is often stored in your body, and discharged when you come in contact with an object of a different potential.
   Perform the procedure below (removal of chassis cover) described in this section only at an ESD workstation using an antistatic wrist strap. If such a station is not available, you can provide some ESD protection by wearing an antistatic wrist strap and attaching it to a metal ground screw (lug) next to power switch of the chassis.
3. Slide the cover forward and tilt the rear of the cover up as you remove it.
4. Insert the SIM cards into the SIM card slots on the internal wireless modems. Each modem has a graphic line depicting the correct SIM card orientation.
5. Verify that the SIM cards fit into their holders properly and then replace the cover by reversing what you did in step 3. Align the “teeth” at the front of the unit first.
6. Attach the screws and then the power cord.

Modem Numbering

The MVPGSM-2 modem numbering corresponds to the printing on the front panel. You may also tell the numbering by the wires routed to the antennas.

Cabling Procedure

Cabling involves connecting the MultiVOIP GSM to power, your LAN and a serial console.

1. Connect a power cord to the transformer and to a live AC outlet, and then attach the barrel connector to the back power receptacle of the MultiVOIP GSM.
2. Connect the MultiVOIP GSM to a PC by using a RJ-45 (male) to DE-9 (female) cable. Plug the RJ-45 end of the cable into the **CONSOLE** port of the MultiVOIP GSM and the other end into the PC serial port.
3. Connect a network cable to the **WAN** connector on the back of the MultiVOIP GSM. Connect the other end of the cable to your network.
4. Attach an antenna to both channel connectors on either side of the MultiVOIP GSM.

   Caution: A separation distance of at least 20 cm (8 inches) must be maintained between all transmitting antennas and the body of the user or nearby persons. This device is not designed for or intended to be used in portable applications within 20 cm (8 inches) of the body of the user.
5. Turn on power to the MultiVOIP GSM by placing the ON/OFF switch on the back panel to the ON position. Wait for the **BOOT** LED on the MultiVOIP GSM to go off before proceeding. This may take a few moments.
6. Proceed to the *Software Installation* chapter to load the MultiVOIP GSM software.

GSM Instructions

Step 1 – Activate Your Wireless Accounts

Select a wireless network provider and follow their directions to activate your account and receive your SIM cards.

Phone Numbers

Each channel will have its own unique phone number. The phone number may simply be given to you by your wireless service provider or it may be on the SIM card or both. Wireless provider implementations may vary.

Step 2 – Check Signal Strength

Turn the unit on and verify that the Power LED is lit and that the Boot sequence is finished (Boot LED is no longer active), then wait for the Link Status (LS) LED to show that the MVPGSM channel is registered on the wireless network (flashing 75 ms on and 3 seconds off). Once registered, the Signal Strength LEDs should be referenced for the strength of signal in its current location.

Caution: Before final placement or mounting, ensure that the wireless signal strength is strong enough for the chosen area. Signal strength information is found in the Front Panel LEDs section of Chapter 1. Finding a location with the strongest signal strength is desirable.

Note about Wireless Bands

The wireless units inside the MVPGSM support quad band frequencies (850/1900/900/1800 MHz). In reality, these operate like dual, dual-band devices. In other words, they can be configured for 850/1900 or 900/1800 MHz. They do not auto-seek the local area frequency.

Should you need to change the default band setting, follow the instructions below:

Go to **Configuration | Wireless Setup | Interface Parameters**.

Type **AT+WMBS=<*Band*><*Param*>** into the *Modem InitString1* box.

For **<Band>**, enter the option you desire:

4 = Dual-band mode 850/1900MHz

5 = Dual-band mode 900/1800MHz

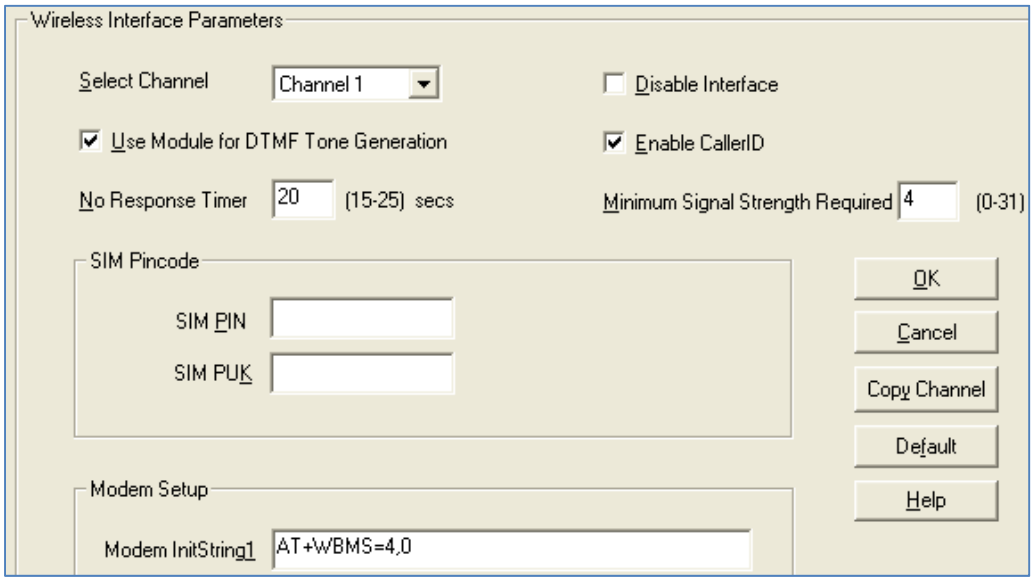
For **<Param>**, enter the option you desire:

0 = Modem must be reset in order to use the specified band(s). This is the default.

1 = Modem restarts immediately using the specified band(s).

Example: **AT+WMBS=4,0**

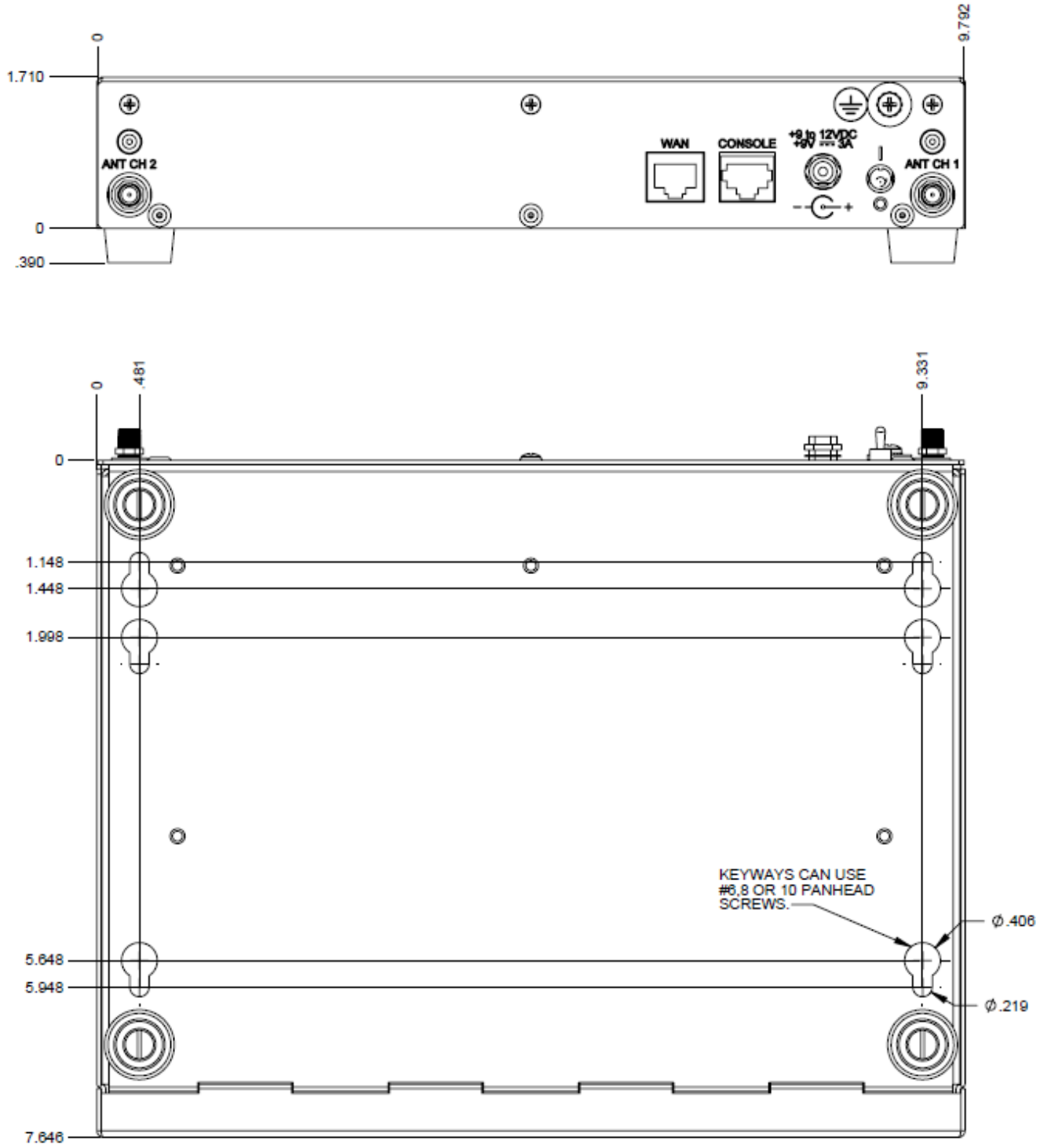
Click the Copy Channel button and both channel boxes, then click the Copy button.
Click OK on the Interface Parameters screen.
Go to Save Setup | Save & Reboot and click OK.



The MVPGSM will save the settings and reboot. Once the MVPGSM has rebooted, check to make sure the wireless units can connect. You may now go to **Configuration | Wireless Setup | Interface Parameters** and remove the *AT+WMBS* command from the *Modem InitString1* and select the 'copy to all' button to make the change for all channels and then click the Copy button. Then you can click OK on the Interface Parameters screen and use the **Save & Reboot** option one more time. There is no need to have the MVPGSM send the +WMBS command every time it initializes an internal modem, so we are reversing the process detailed above.

Mounting Instructions

When not used as a tabletop device, the bottom panel of the MVPGSM has six keyed slots for versatility of mounting. The dimensions (in inches) are provided below.



Chapter 3 – MultiVOIP Software

Introduction

Configuring software for your MultiVOIP GSM entails three tasks:

Loading the software onto the PC (this is “Software Installation” and is discussed in this chapter).

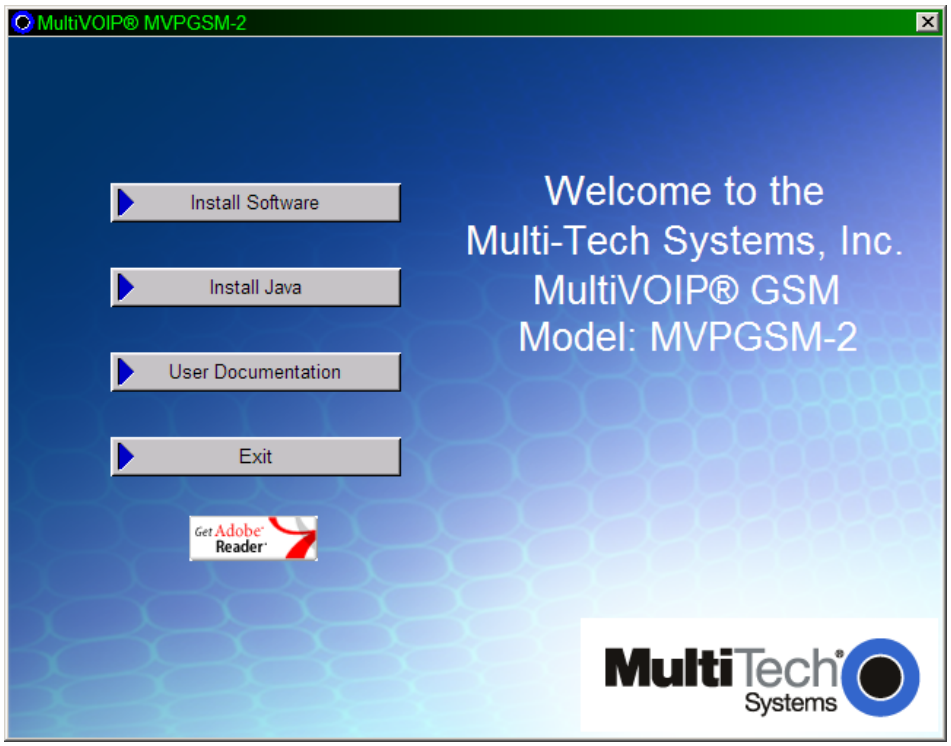
Setting values for telephony and IP parameters that will fit your system (details are in Chapter 4).

Establishing “phonebooks” that contain the various dialing patterns for VOIP calls made to different locations (a detailed discussion of this is found in Chapter 5).

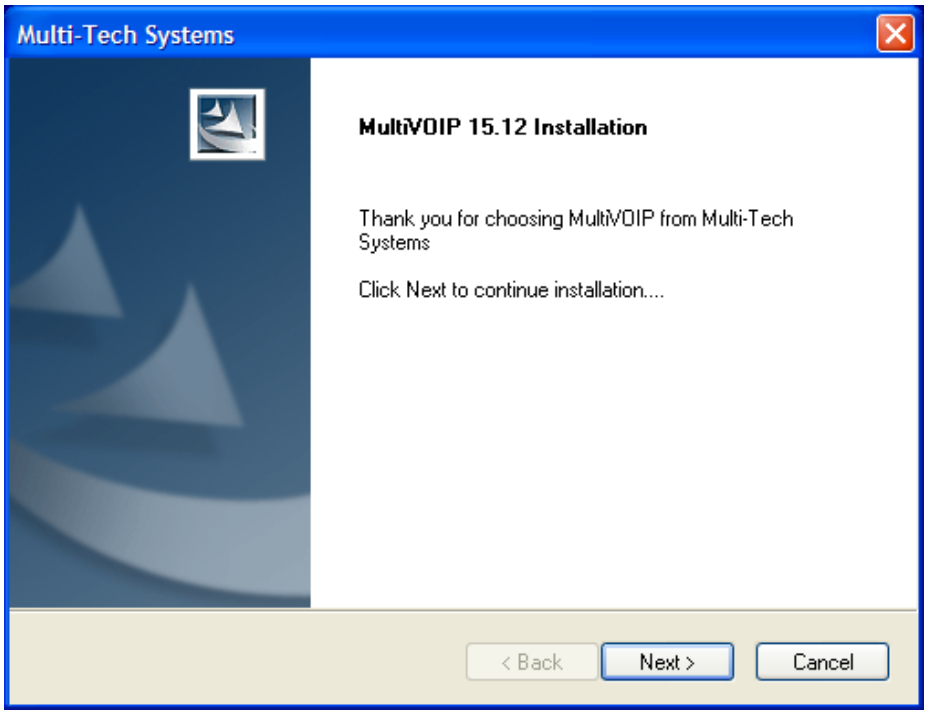
Installing the MultiVOIP GSM Software

The software loading procedure does not present every screen or option in the loading process. It is assumed that someone with a thorough knowledge of Windows and the software loading process is performing the installation.

1. Be sure that your MultiVOIP GSM has been properly cabled and that the power is turned on.
2. Insert the MultiVOIP GSM CD into your CD-ROM drive. The CD starts automatically. It may take a few moments for the Multi-Tech CD installation window to display.

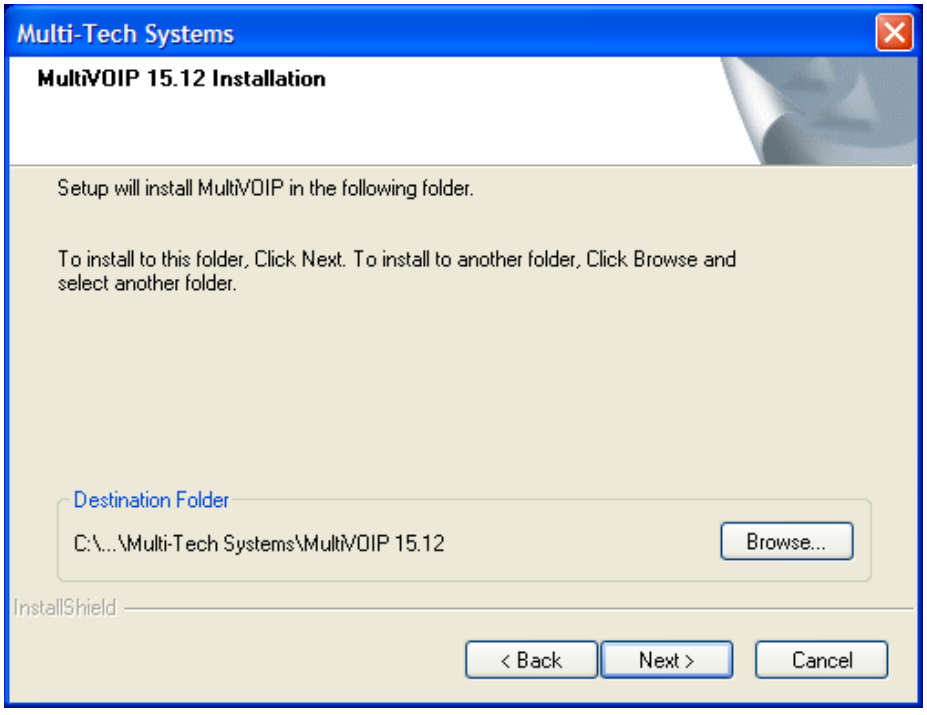


3. When the Multi-Tech Installation CD dialog box appears, click the **Install Software** button.



Press **Enter** or click **Next** to continue.

4. Follow the on-screen instructions to install your MultiVOIP GSM software. The first screen asks you to choose the destination for the MultiVOIP GSM software.


Multi-Tech Systems
MultiVOIP 15.12 Installation
Setup will install MultiVOIP in the following folder.
To install to this folder, Click Next. To install to another folder, Click Browse and select another folder.
Destination Folder
C:\...\Multi-Tech Systems\MultiVOIP 15.12
Browse...
InstallShield
< Back
Next >
Cancel

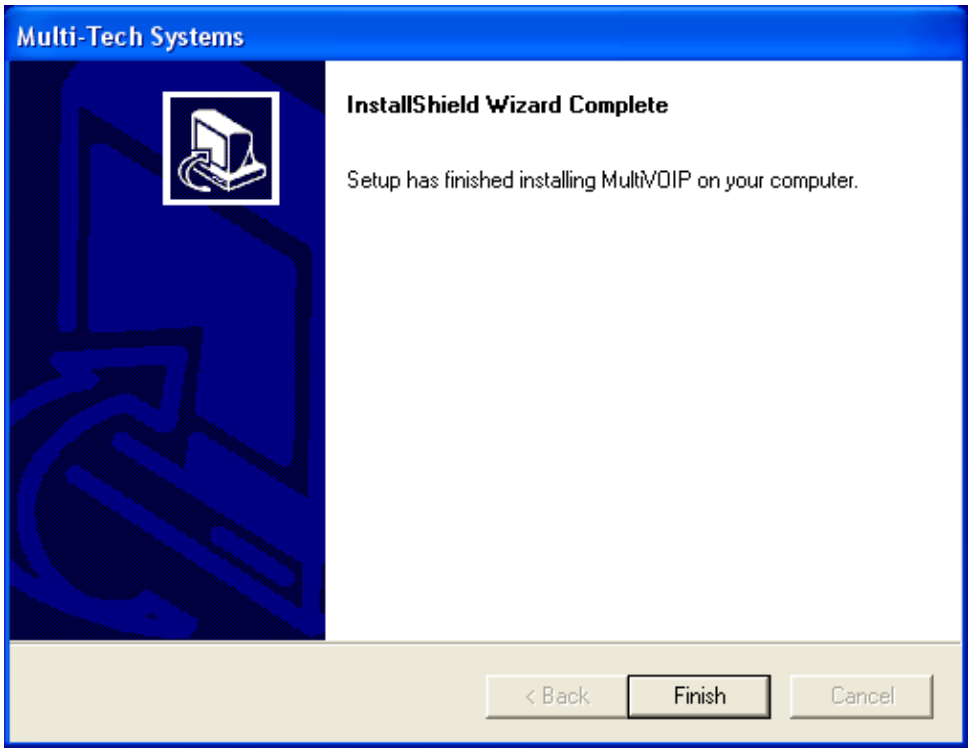
Choose a location and click **Next.**

5. At the next screen, you must select a program folder location for the MultiVOIP GSM software program icon. Click **Next**. Transient progress screens will appear while files are being copied.

6. On the next screen you can select the COM port that the command PC will use when communicating with the MultiVOIP GSM unit. After software installation, the COM port can be re-set in the MultiVOIP GSM Software (from the sidebar menu, select **Connection | Settings** to access the **COM Port Setup** screen or use keyboard shortcut Ctrl + G).

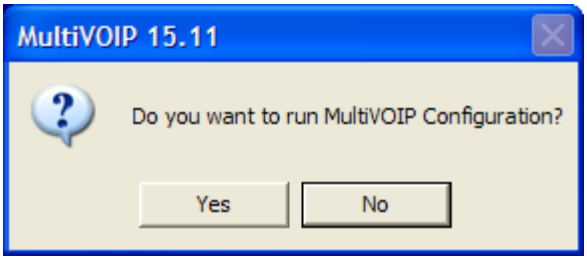
   Note: If the COM port setting made here conflicts with the actual COM port resources available in the command PC, the "Error in Opencomm handle" message will appear when the MultiVOIP GSM program is launched. If this occurs, you must reset the COM port.

7. A completion screen will appear.



   Click **Finish**.

8. When setup of the MultiVOIP GSM software is complete, you will be prompted to run the MultiVOIP GSM software to configure the VOIP.



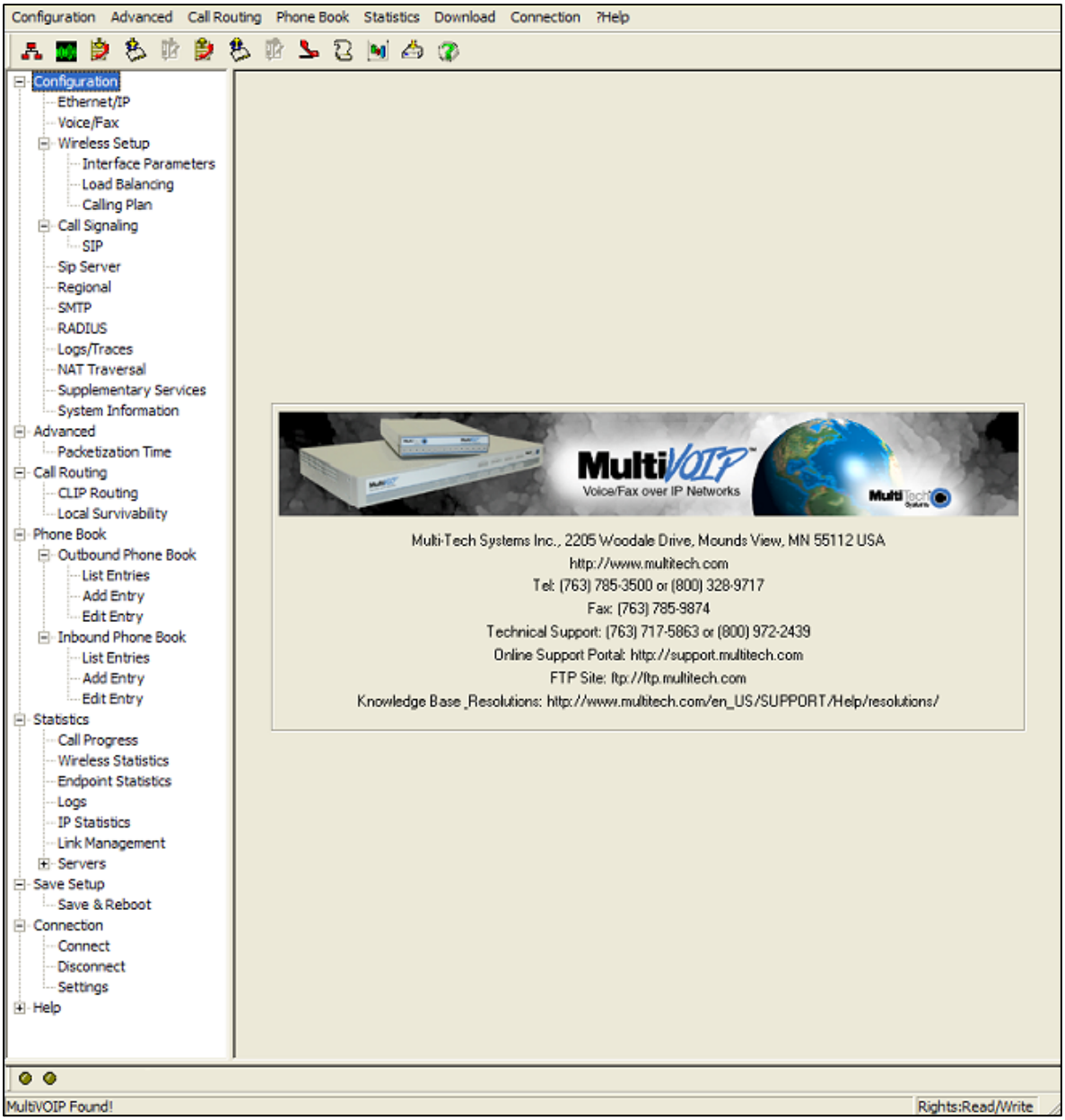
Software installation is now complete.

Basic Setup

With the software now installed, you are ready to get your MultiVOIP GSM set up and working. There are a few necessary settings that need to be entered in the configuration software to achieve this and they are noted in the *action* lists for the categories below. The following chapters will cover all aspects in detail, but here we will cover the basic configuration needed to start VOIP communications. Below you will find the list of categories requiring information to be set before VOIP communication will be ready.

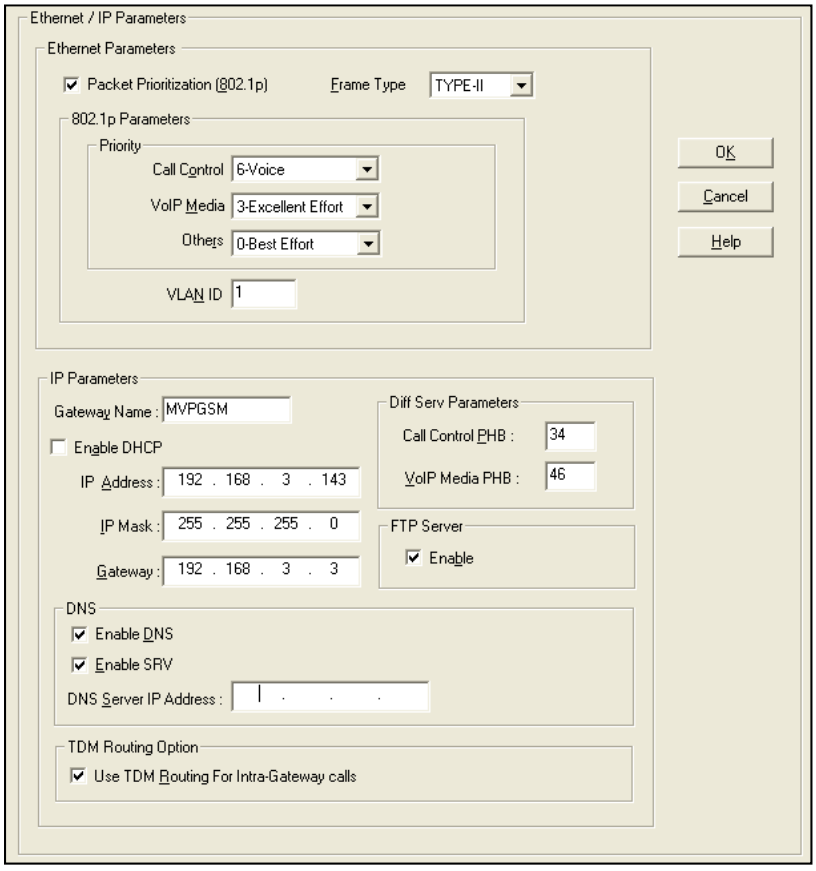
- ⇒ **Ethernet/IP**
- ⇒ **Voice/Fax**
- ⇒ **Wireless Setup**
- ⇒ **Call Signaling**
- ⇒ **Regional**
- ⇒ **Phone Book**

This setup process must be followed by a **Save & Reboot** for the changes to take effect.



Ethernet/IP

A unique LAN IP address is required for the MultiVOIP GSM unit as well as a subnet mask and Gateway IP for minimal functionality. Other settings in this category pertain to specific features and protocols that can be used, but are not necessary for basic operation. Details for all settings are provided in Chapter 4.

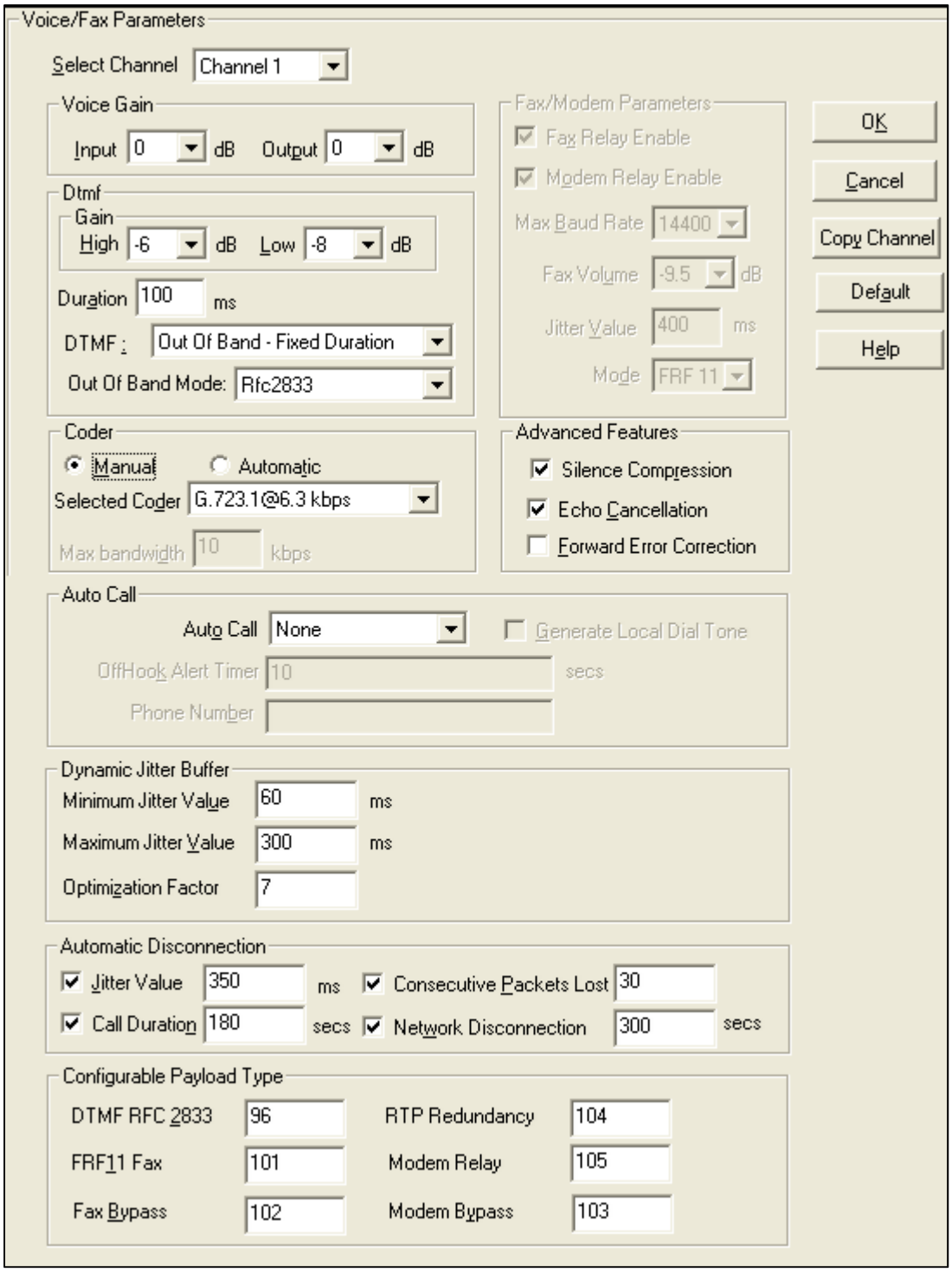


Actions:

- Select Packet Prioritization if used
  - Set 802.1p Priority Parameters as needed
    - The Priority levels can be from 0 – 7, where 0 is lowest priority (details in Chapter 4)
    - VLAN ID identifies a virtual LAN by a number (1 to 4094)
- Set the Frame Type to match the network that the MultiVOIP GSM is attached to
  - TYPE II or SNAP
- Enter Gateway Name
- Check to enable DHCP if used
- Enter IP Address for the MultiVOIP GSM unit (default is 192.168.3.143)
- Enter Subnet IP Mask for the MultiVOIP GSM unit
- Enter Gateway IP
- Enable DNS if desired
  - Enter DNS Server IP Address
- Enable SRV support if needed
- Diff Serv Parameters are for routers that are Diff Serv compatible
  - Setting both values to 0 effectively disables Diff Serv
- FTP Server Enable is only needed for firmware and software updates to the MultiVOIP GSM
- TDM Routing can be used if necessary

Voice

The individual channels must be set up before use. The Copy Channel button can save a lot of time during this step if channels are to be set with the same parameters. Some options should be noted for future changes if necessary, but the defaults are likely to work without adjustment.



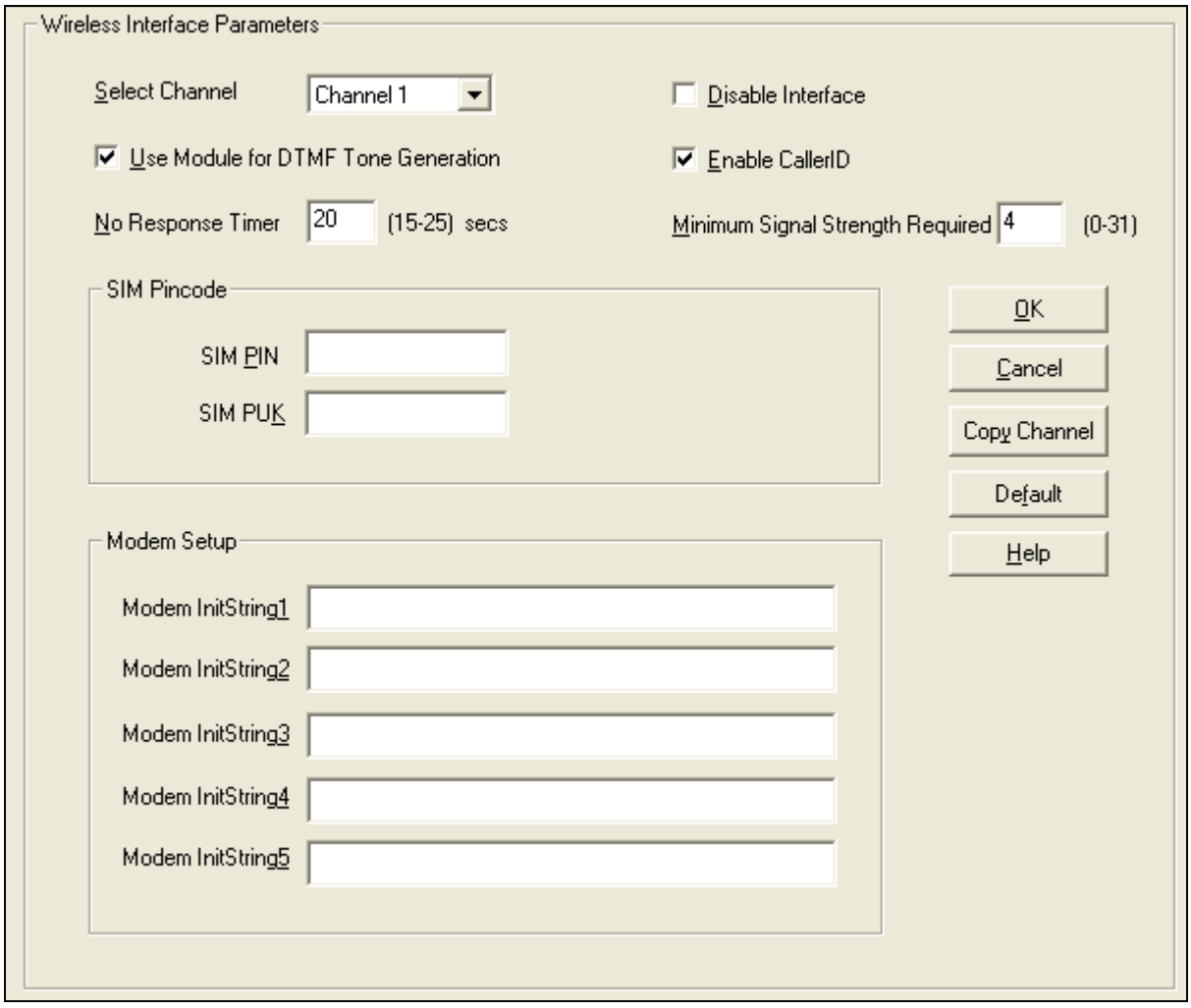
Actions:

- Select Channel
  - Choose channel parameters:
    - Fax and modem parameters are not available at this time
    - Adjusting the Voice Gain and DTMF *should not be done* as it may adversely affect voice and DTMF quality and recognition
    - Select a Coder or allow Automatic negotiation
    - Advanced Features
      - Silence Compression, when enabled, will not send IP packets during times of silence
      - Echo Cancellation removes echo to improve voice quality
      - Forward Error Correction allows some bad packets to be recovered
    - Choose Auto Call settings
      - For automatically calling a remote VOIP without dialing (details in Chapter 4)
    - Change Dynamic Jitter values if necessary (details in Chapter 4)
    - Select any Automatic Disconnection options needed to ensure lines are not left "open"
    - Configurable Payload Types are best left at their defaults.
  - The Copy Channel button is available for easily transferring these settings to the other channel
  - The Default button will return settings to factory default settings

Wireless Setup

Wireless Interface

The Wireless Interface Parameters are the settings for the GSM connection. The Copy Channel button can save a lot of time during this step if channels are to be set with the same parameters.

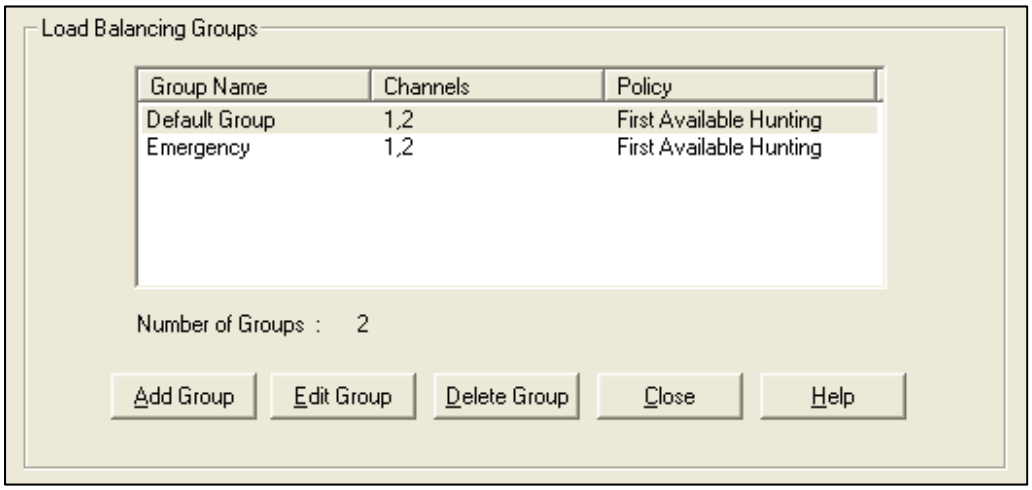


Actions:

- Select Channel
  - Select the channel you want to edit
- Disable Interface
  - Check box: Enable or Disable Interface
- Use Module for DTMF and Tone Generation
  - Check box: Enable or Disable DTMF and Tone generation (this is used in conjunction with out of band DTMF)
- Enable Caller ID
  - Check box: Enable or Disable Caller ID (this determines if Caller ID information coming in from the wireless network is to be sent over the IP network)
- No Response Timer
  - Internal timer for command response time. Range 15-25 seconds; default is 20
- Minimum Signal Quality Required
  - Range 0-31
- SIM Pincode
  - SIM PIN
    - Enter SIM PIN number (this is only necessary in cases where the PIN is required on power-up)
  - SIM PUK
    - Enter SIM PIN Unlock Key number (this would only be needed in conjunction with the above PIN number where 3 incorrect attempts lock an account)
- Modem Setup
  - Modem InitString 1-5
    - Enter initialization strings (if necessary, internal modem commands can be added)
- Copy Channel (button)
  - Copy settings to another channel
- Default (button)
  - Reset unit to Factory Default Settings

Load Balancing

For those installations without unlimited use plans, load balancing can be enabled to keep the first channel from being over-used while the other channel sees less traffic.

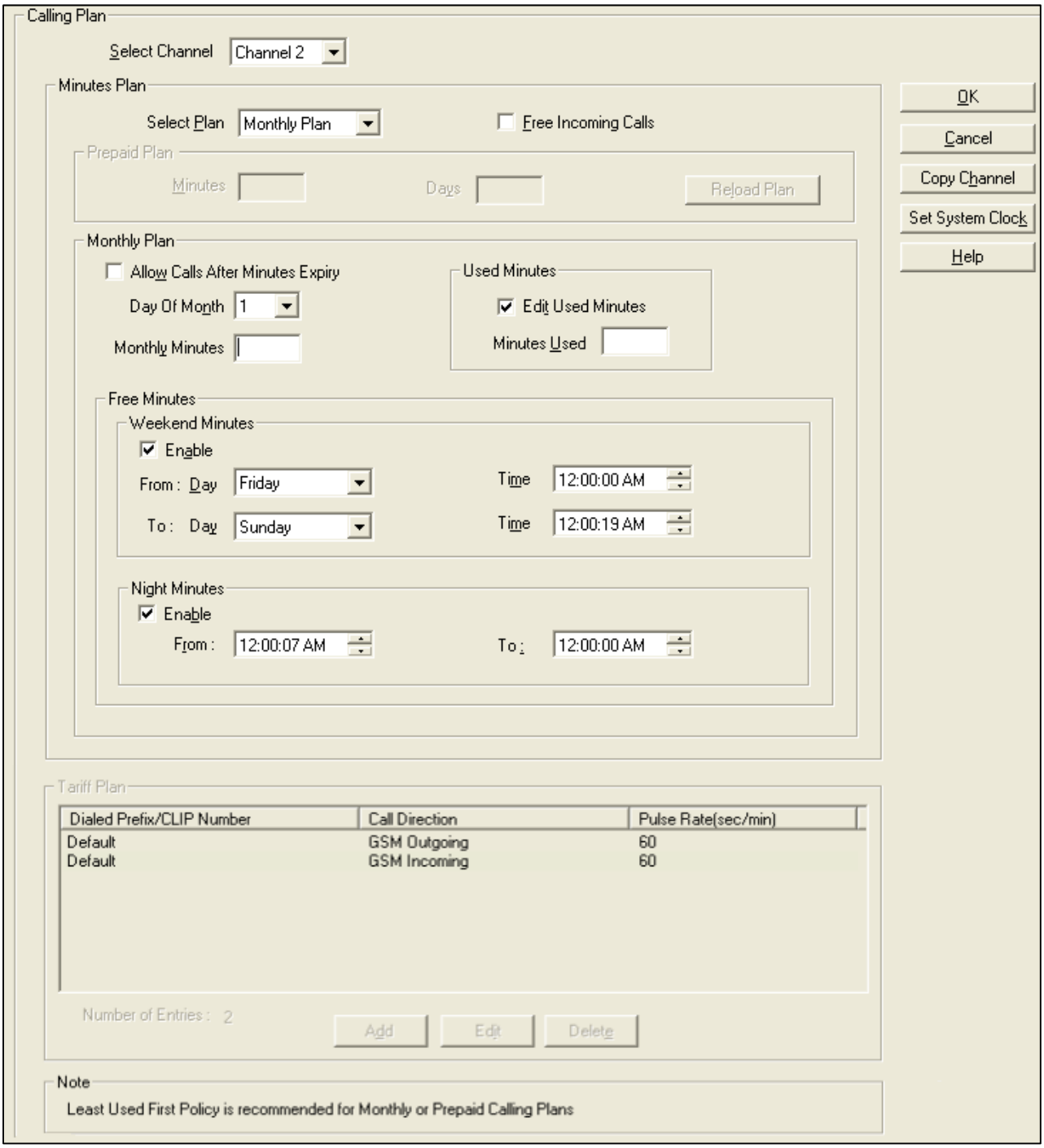


Actions:

- Group Name
  - By default, the MVPGSM has two groups available (Default and Emergency). You may Add or Edit as needed
- Channels
  - This lists the channels that are used by the group
- Policy
  - This shows a brief statement of the policy in place for the group
- Number of Groups
  - This is a total of all the created load balancing groups
- Add Group (button)
  - This will open the add/Edit group screen for adding a new group
- Edit Group (button)
  - This will open the add/Edit group screen for editing the selected group
- Delete Group (button)
  - This will permanently delete the selected group
- Close (button)
  - The close button will close this screen and return you to the next highest option

Calling Plan

If you are using the *Least Used First* (LUF) load balancing option or a SIM designated as prepaid or a monthly plan, this Calling Plan screen needs to be completed with the specifics for each SIM with a calling plan.



Actions:

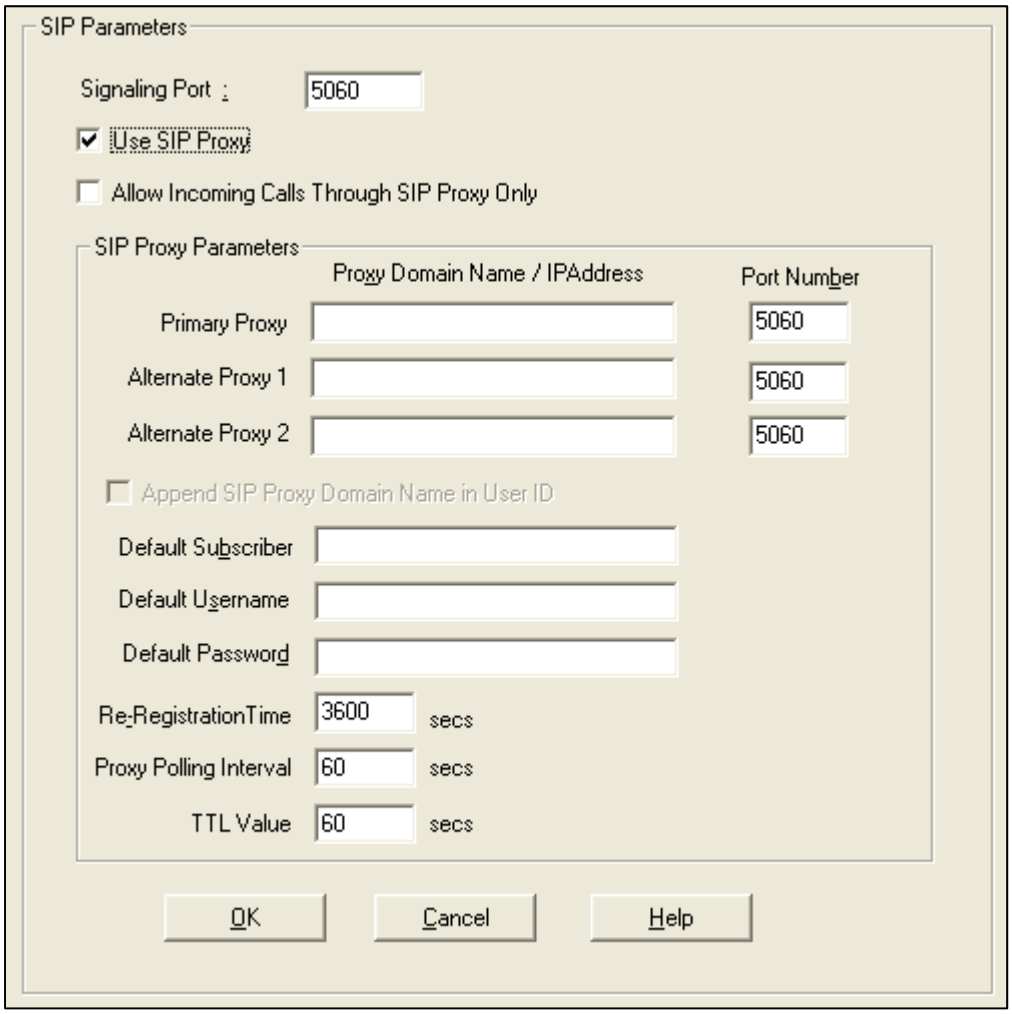
- Select Channel
  - Select the channel you want to edit (set parameters for every channel that has a SIM)
- Minutes Plan
  - Select Plan
    - Unlimited
    - Prepaid Plan
    - Monthly Plan
  - Free Incoming Calls
    - Check this box if the calls received by the MVPGSM on this channel are free
  - Prepaid Plan
    - Minutes
      - Enter the minutes available for this channel (SIM)
    - Days
      - Enter the days available for this channel (SIM)
    - Reload Plan (button)
      - Clicking the Reload Plan button will reset all used minutes to zero
  - Monthly Plan
    - Allow calls after minutes expiry (checkbox)
      - Check this box if you want this channel to continue being used after the minutes have been used up
    - Monthly Minutes
      - Enter the total monthly minutes allowed on this channel (SIM)
    - Day of Month
      - Enter the recharge day of the month for this channel (SIM)
    - Used Minutes
      - Edit Minutes Used
        - If you want to override the calculated 'Minutes Used' field, check this box and the Minutes Used will be set to '0' and you can enter a new value
      - Minutes Used
        - Each time this page is accessed, this will display the minutes used so far on this channel (SIM)
    - Free Minutes
      - Weekend Minutes
        - Enable (checkbox)
          - If this channel has free weekend minutes, check this box
        - From:
          - Enter the Day & Time when free weekend minutes begin
        - To:
          - Enter the Day & Time when free weekend minutes end
      - Night Minutes
        - Enable (checkbox)
          - If this channel has free nighttime minutes, check this box
        - From:
          - Enter the Time when free night minutes start
        - To:
          - Enter the Time when free night minutes end

Calling Plan (continued)

- Tariff Plan
  - Dialed Prefix/CLIP Number
    - The dialed prefix (matching CLIP pattern) of a GSM call for which the tariff is applicable
  - Call Direction
    - Will display the direction of the call (Outgoing or Incoming)
  - Pulse Rate
    - Displays the pulse rate
  - Number of Entries
    - Displays the total number of entries in the Tariff Plan
  - Add/Edit (buttons)
    - Add a new plan or edit an existing plan
  - Delete (button)
    - This will permanently delete the selected plan
- Copy Channel (button)
  - This will copy all the information entered for this channel and allow you to copy to the other channel
- Set System Clock (button)
  - Set the system clock to the current date, day and time for accurate use of the Calling Plan settings

Call Signaling

The MultiVOIP GSM utilizes the SIP protocol for communication with other VOIP units. Additional details for all settings are found in later chapters.

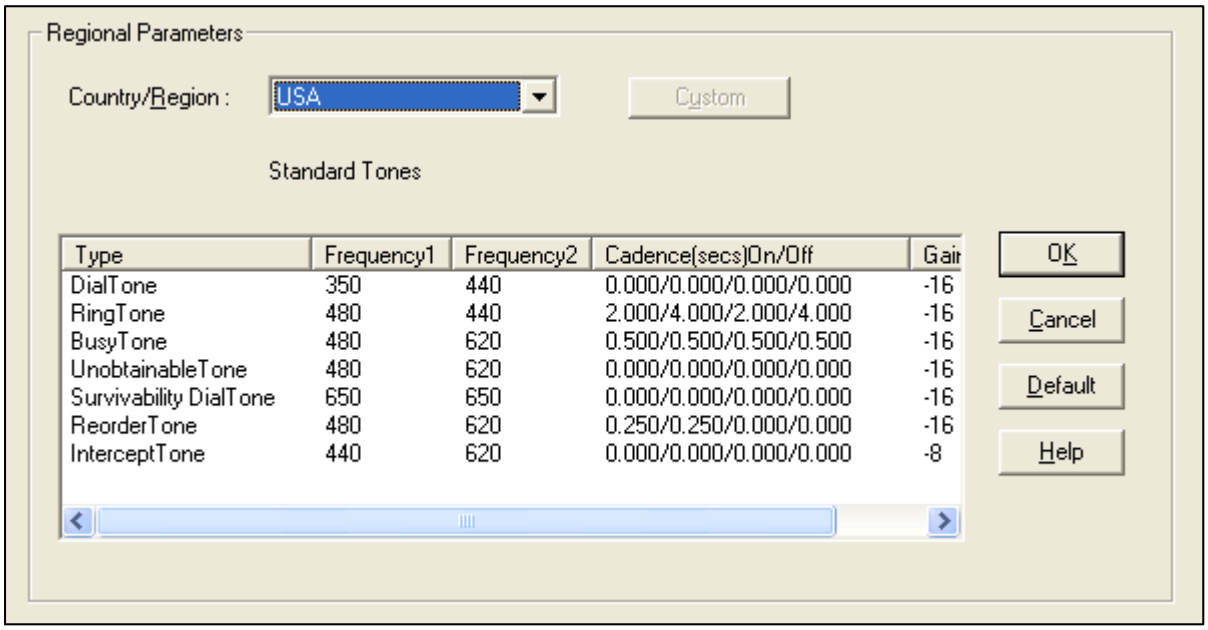


Actions:

- Configure the Call Signal type
  - SIP
    - Signaling Port (default is 5060)
    - Use SIP Proxy (enable to work with a proxy server)
    - Allow Incoming Calls Through SIP Proxy Only
    - SIP Proxy Parameters
      - Enter information for Primary and any Alternate Proxy servers
      - Append SIP Proxy Domain Name in User ID
      - Enter Subscriber number, Username and Password
      - Re-Registration Time (in seconds)
      - Proxy Polling Interval (time between proxy server connect attempts)
      - TTL Value (in seconds)

Regional

Select the country or region that the MultiVOIP GSM unit will operate in, or use the custom option if the available settings are not adequate.

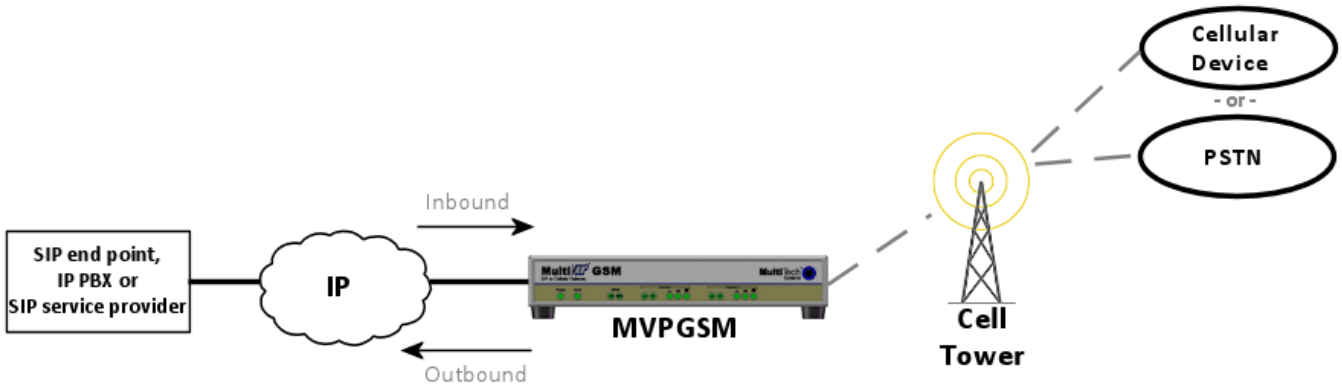


Actions:

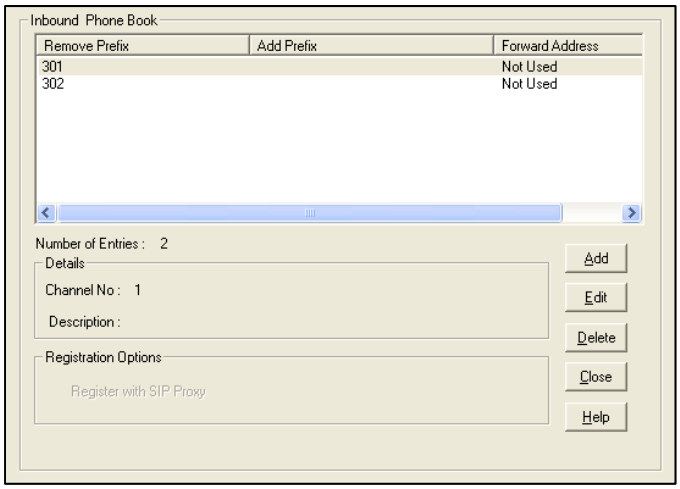
- Select the choice that matches the location of the MultiVOIP GSM from the Country/Region field
  - If there is not a selection to fit your needs, you may select Custom and set the tones manually

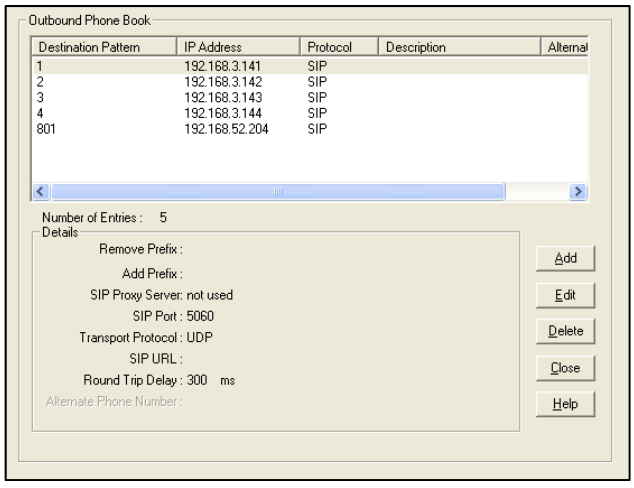
Phone Book

Without a populated phone book, the VOIP unit is unable to translate call traffic. You will need the information for both a local and any remote sites that are to be used. Detailed descriptions and examples are available in chapter 5. To better understand the meaning behind 'Inbound' and 'Outbound' please see the graphic below.



Inbound and Outbound directions





Actions:

- Select Outbound Phone Book
  - Select Add Entry
  - Accept Any Number: may be selected to allow any phone number from the wireless network to initiate a SIP call to the IP address of the SIP end point or load balancing group listed below
  - Destination Pattern: When a call from the wireless network is received, the subsequent digits received from the wireless side are used to match the destination pattern and route the call to the SIP end point with the IP address listed below
  - Remove Prefix: enter any access digits (e.g. a number to get out of a PBX system) or area codes to be taken off in this field – this manipulates the number dialed by the wireless-side caller before sending the digit string to the SIP end point of the IP address below
  - Add Prefix: any digits that need to be added to make a call appear as local should be put in this field – this manipulates the number dialed by the wireless-side caller before sending the digit string to the SIP end point of the IP address below
  - Enter the IP address of the SIP end point where the phone number is to be routed
  - Protocol type
    - SIP:
      - Select Call Signaling Port, Transport Protocol, Proxy and URL if needed
  - The *Advanced Button* will allow routing to an Alternate IP Address to be entered for times when the primary does not respond within the allotted time
- Select Inbound Phone Book
  - Select Add Entry
  - Accept Any Number will allow any number received from a SIP end point to be routed to the selected channel below
  - Enter any access digits followed by the local calling code in the Remove Prefix field. Incoming SIP calls that match these digits will be routed to the selected channel below and have those digits removed from the incoming number
  - Enter any digits needed to access an outside line in the Add Prefix field. These will be added to the prefix of the SIP inbound call
  - Select *Default Group* or one of the groups added on the Load Balancing page in the Channel Number field to have the VOIP use channels efficiently (this is necessary for load balancing), otherwise you can select a specific channel for the call to go out on
  - Add a description if you like
  - Call Forward may be set up (details available in Chapter 5)
  - Select Registration Option
- Repeat the Phone Book steps for any additional entries needed

Save & Reboot

Any time that you change settings on the VOIP unit, you must choose the **Save & Reboot** option; otherwise all changes made will not take affect and be lost completely when the MultiVOIP GSM is reset, shutdown or you exit from the management screen.

Chapter 4 – Configuring Your MultiVOIP GSM

Introduction

There are two methods of configuring your MultiVOIP GSM; one is through a web interface, and the other is through the Windows software interface. There are several necessary parameters that must be set for the MultiVOIP GSM unit to operate properly, with some additional settings that are optional. You must know the IP address that will be used, the IP mask, the Gateway IP, and the Domain Name Server information. The MultiVOIP GSM is generally configured locally at first, but changes to this initial configuration can be done locally or remotely. Local configuration is done through a connection between the "Console" port of the MultiVOIP GSM and the COM port of the computer; the MultiVOIP GSM configuration software is used for this.

This chapter will explain the setup portion of the software pertaining to the list below, while Chapter 5 will cover the Phone Book setup and Chapter 6 will discuss the Statistics options and overall maintenance of the MultiVOIP GSM.

Software Categories Covered in This Chapter

- **Ethernet/IP**
- **Voice/Fax**
- **Wireless Setup**
  - **Interface Parameters**
  - **Load Balancing**
  - **Calling Plan**
- **Call Signaling**
  - **SIP**
- **SIP Server**
- **Regional**
- **SMTP**
- **RADIUS**
- **Logs/Traces**
- **NAT Traversal**
- **Supplementary Services**
- **Call Routing**
  - **CLIP Routing**
  - **Local Survivability**
- **Save Setup**
- **Connection**
  - **Settings**

How to Navigate Through the Software

The MultiVOIP GSM software is launched from the Windows Start button and is found in the All Programs area under the title of MultiVOIP GSM *x.xx* (where x represents version number). The top option is "Configuration" – choose this.

Within the software, there are several ways to arrive at the parameter that you want to use: through the left-hand panel, from the drop-down menu, clicking a taskbar icon (if available) or a keyboard shortcut (if available). Once the initial settings are entered, you may choose to configure the MultiVOIP GSM through a Web browser instead.

Web Browser Interface

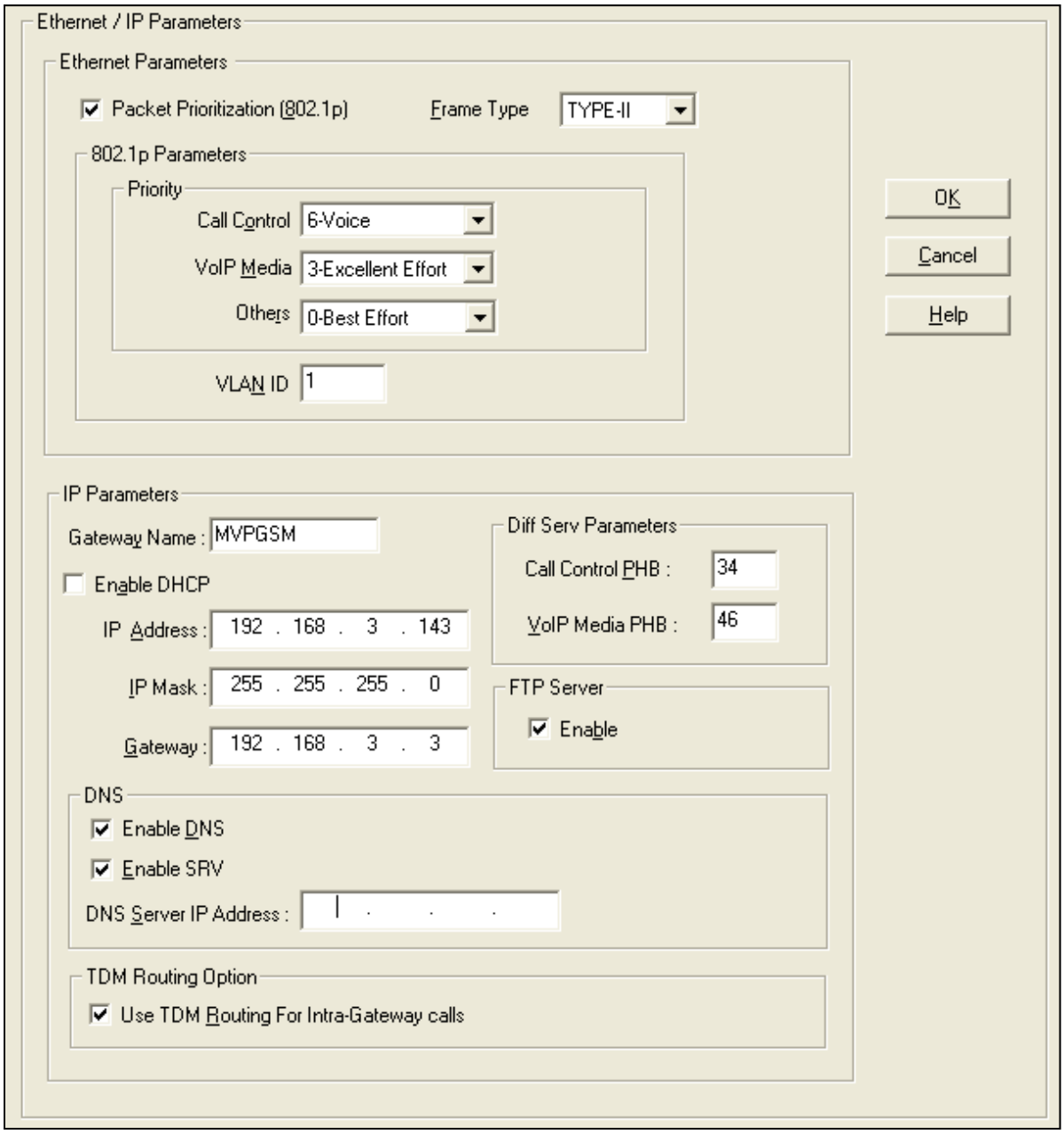
The MultiVOIP GSM web browser interface gives access to the same commands and configuration parameters as are available in the MultiVOIP GSM Windows interface except for logging functions. When using the web browser interface, logging can be done by email (the SMTP option).

Set up the Web Browser interface (Optional). After an IP address for the MultiVOIP GSM unit has been established, you can choose to configure the unit by using the MultiVOIP GSM web browser interface. If you want to do configuration work using the web browser interface, you must first set it up:

- Set IP address of MultiVOIP GSM unit using the MultiVOIP GSM Configuration program (the Windows interface). **The default IP address is 192.168.3.143**.
- Save Setup in Windows interface.
- Close Windows interface.
- Install Java program from MultiVOIP GSM product CD (on first use only).
- Open web browser.
- Browse to IP address of MultiVOIP GSM unit.
- If a username and password have been set up, enter them when prompted.
- Set the browser to allow pop-ups. The MultiVOIP GSM Web interface makes use of pop-up windows.
- The configuration screens in the web browser will have the same content as their counterparts in the software; only the presentation differs.

Ethernet/IP

This section covers the Ethernet settings needed for the MultiVOIP GSM unit. In each field, enter the values that fit the network to which the MultiVOIP GSM will be connected to. For many of the settings, the default values will work best – try these settings first unless you know you definitely need to change a parameter.



The **Ethernet/IP Parameters** fields are described in the tables and text passages below.

Ethernet/IP Parameter Definitions		
Field Name	**Values**	**Description**
Ethernet Parameters		
Packet Prioritization (802.1p)	checkbox	Select to activate prioritization under 802.1p protocol (described below).
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.
802.1p	A draft standard of the IEEE about data traffic prioritization on Ethernet networks. The 802.1p draft is an extension of the 802.1D bridging standard. 802.1D determines how prioritization will operate within a MAC-layer bridge for any kind of media. The 802.1Q draft for virtual local-area-networks (VLANs) addresses the issue of prioritization for Ethernet networks in particular. 802.1p enacts this Quality-of-Service feature using 3 bits. This 3-bit code allows data switches to reorder packets based on priority level. The descriptors for the 8 priority levels are given below. 802.1p PRIORITY LEVELS: *LOWEST PRIORITY* 1 – **Background**: Bulk transfers and other activities permitted on the network, but should not affect the use of network by other users and applications. 2 – **Spare**: An unused (spare) value of the user priority. 0 – **Best Effort** (default): Normal priority for ordinary LAN traffic. 3 – **Excellent Effort**: The best effort type of service that an information services organization would deliver to its most important customers. 4 – **Controlled Load**: Important business applications subject to some form of "Admission Control", such as preplanning of Network requirement, characterized by bandwidth reservation per flow. 5 – **Video**: Traffic characterized by delay < 100 ms. 6 – **Voice**: Traffic characterized by delay < 10 ms. 7 - **Network Control**: Traffic urgently needed to maintain and support network infrastructure. *HIGHEST PRIORITY*	
Call Control Priority	0-7	Sets the priority for signaling packets. 0 is lowest priority.
VOIP Media Priority	0-7	Sets the priority for media packets. 0 is lowest priority.
Others (Priorities)	0-7, where 0 is lowest priority	Sets the priority for SMTP, DNS, DHCP, and other packet types. 0 is lowest priority.
VLAN ID	1 - 4094	The 802.1Q IEEE standard allows virtual LANs to be defined within a network. This field identifies each virtual LAN by number.
IP Parameter fields		
Gateway Name	alphanumeric	Descriptor of current VOIP unit to distinguish it from other units in system.
Enable DHCP	checkbox disabled by default	Dynamic Host Configuration Protocol is a method for assigning IP address and other IP parameters to computers on the IP network in a single message with great flexibility. IP addresses can be static or temporary depending on the needs of the computer.
IP Address	*n.n.n.n*	The unique IP address assigned to the MultiVOIP GSM. Default is 192.168.3.143
IP Mask	*n.n.n.n*	Subnetwork address that allows for sharing of IP addresses within a LAN.
Gateway	*n.n.n.n*	The IP address of the device that connects your MultiVOIP GSM to the Internet.

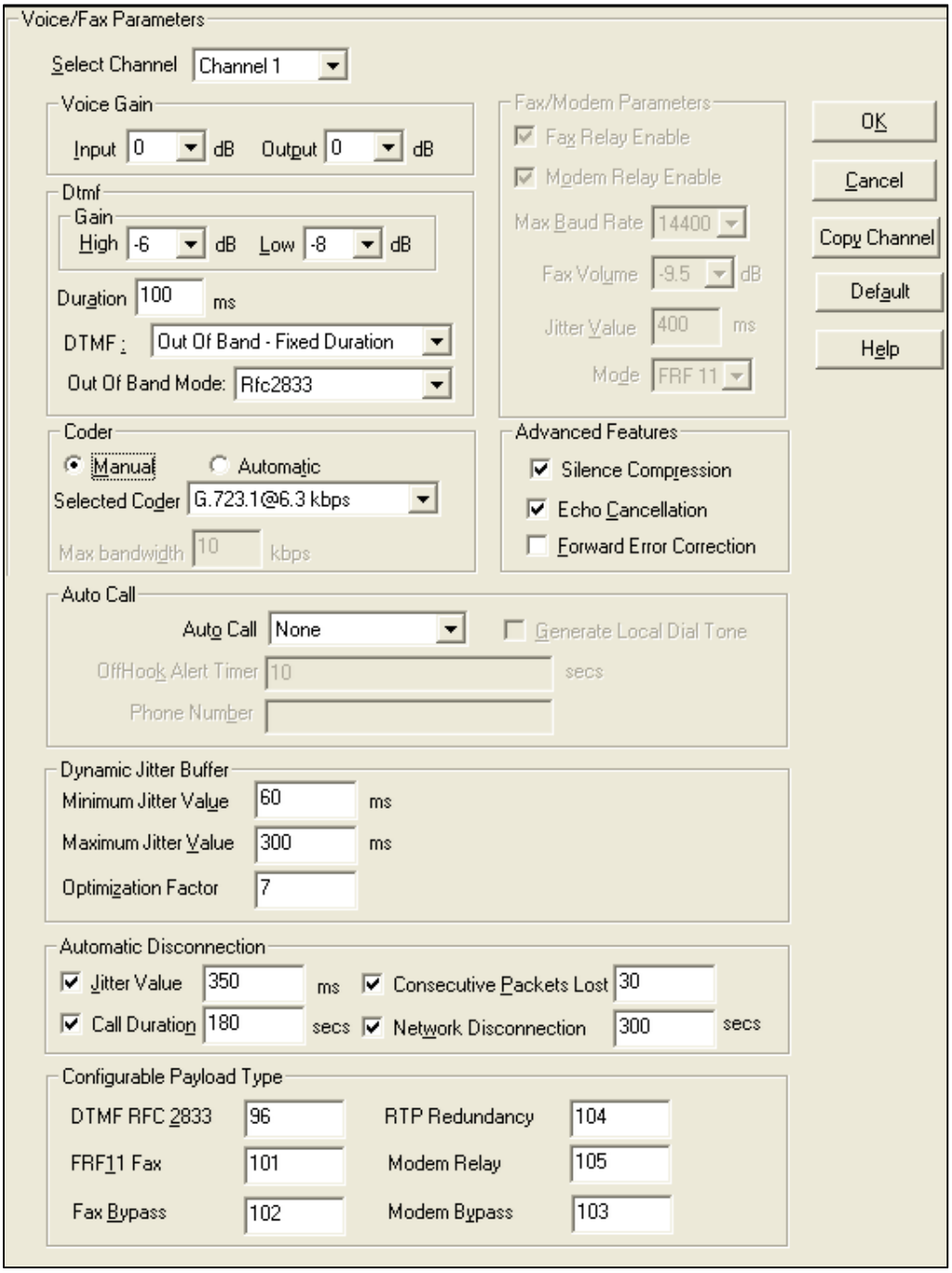
Table is continued on next page…

Ethernet/IP Parameter Definitions (continued)		
Field Name	**Values**	**Description**
Diff Serv Parameter fields	Diff Serv PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by Diff Serv-compatible routers. There are 64 values, each with an elaborate technical description. These descriptions are found in TCP/IP standards RFC2474, RFC2597, and, for present purposes, in RFC3246, which describes the value 34 (34 decimal; 22 hex) for Assured Forwarding behavior (default for Call Control PHB) and the value 46 (46 decimal; 2E hexadecimal) for Expedited Forwarding behavior (default for VOIP Media PHB). Before using values other than these default values of 34 and 46, consult these standards documents and/or a qualified IP telecommunications engineer. To disable Diff Serv, configure both fields to 0 decimal.	
Call Control PHB	0 – 63 default = 34	Value is used to prioritize call setup IP packets. Setting this parameter to 0, in conjunction with VOIP Media PHB below will disable Diff Serv.
VOIP Media PHB	0 – 63 default = 46	Value is used to prioritize the RTP/RTCP audio IP packets. Setting this parameter to 0, in conjunction with Call Control PHB above will disable Diff Serv.
FTP Parameter fields		
FTP Server Enable	checkbox Default = enabled	The MultiVOIP GSM unit has an FTP Server function so that firmware and other important operating software files can be transferred to the VOIP via the network.
DNS Parameter fields		
Enable DNS	checkbox Default = disabled	Enables Domain Name System function where computer names are resolved using a worldwide distributed database.
Enable SRV	checkbox	Enables 'service record' function. Service record is a category of data in the Internet Domain Name System specifying information on available servers for a specific protocol and domain, as defined in RFC 2782. Newer internet protocols like SIP, STUN, H.323, POP3, and XMPP may require SRV support from clients. Client implementations of older protocols, like LDAP and SMTP, may have been enhanced in some settings to support SRV.
DNS Server IP Address	*n.n.n.n*	IP address of specific DNS server to be used to resolve Internet computer names.
TDM Routing Options		
Use TDM Routing for Intra-Gateway Calls	checkbox Default = disabled	Check this box to enable Time-Division Multiplexing routing when calls are within the same gateway.

Voice/Fax

Setting the Voice Parameters. The Voice/Fax section needs to be set for each channel to be used. However, once you have established a set of Voice parameters for a particular channel, you can apply this entire set of Voice parameters to another channel by using the **Copy Channel** button and its dialog box. To copy a set of Voice parameters to both channels, select "Copy to All" and click **Copy**.

The majority of the settings should be left at their default settings as changes may introduce problems with audio quality. In each field, enter the values that fit your particular setup.



The Voice/FAX Parameters settings are described in the tables below.

Voice/Fax Parameter Definitions		
Field Name	**Values**	**Description**
Default	(button)	When this button is clicked, all Voice/FAX parameters are set to their default values.
Select Channel	number	Channel to be configured is selected here.
Copy Channel	(button)	Copies the Voice/FAX attributes of one channel to another channel.
Voice Gain	--	Signal amplification (or attenuation) in dB.
Input Gain	+31dB to –31dB	Modifies audio level entering voice channel before it is sent over the IP network to the remote SIP device. The default & recommended value is **0 dB**.
Output Gain	+31dB to –31dB	Modifies audio level from IP to the voice channel. The default and recommended value is **0 dB**.
DTMF Gain	--	The **DTMF Gain** (Dual Tone Multi-Frequency) controls the volume level of the DTMF tones sent out for Touch-Tone dialing. Changing the DTMF gain settings affect the detection of DTMF tones, which can lead to the unit not detecting the tones incorrectly or not at all.
DTMF Gain, High Tones	+3dB to -31dB & "mute"	Default value: **-4 dB**. Generally, this setting should not be changed as it affects the DTMF detection procedure. Altering this value can degrade the performance of the unit.
DTMF Gain, Low Tones	+3dB to -31dB & "mute"	Default value: **-7 dB**. Generally, this setting should not be changed as it affects the DTMF detection procedure. Altering this value can degrade the performance of the unit.
DTMF Parameters		
Duration (DTMF)	60 – 3000 ms	When **DTMF: Out of Band – Fixed Duration** is selected, this setting determines how long each DTMF digit 'sounds' or is held. Default = 100 ms.
DTMF In/Out of Band	Inband, OoB – Fixed Duration, OoB – As Received	When **DTMF Out of Band – Fixed Duration** is selected, the MultiVOIP GSM detects and regenerates DTMF tones as set in the Duration field. When **DTMF Out of Band – As Received** is selected, the Duration field will be grayed out as the network will determine the duration for DTMF generation. **DTMF Inband** is listed as an option, but generally should not be used as it may create problems detecting digits and AT command dial strings.
Out of Band Mode	RFC 2833, SIP Info	**RFC2833 method**. Uses RTP mode defined in RFC 2833 to transmit the DTMF digits. **SIP Info method**. Generates DTMF tones on the telephony call leg. The SIP INFO message is sent along the signaling path of the call. You must set this parameter per the capabilities of the remote endpoint with which the VOIP will communicate. The RFC2833 method is the more common of the two methods.
FAX Parameters – This section is grayed-out as Fax is not supported by the wireless channels		
Fax Enable		
Modem Relay Enable		
Max Baud Rate (Fax)		
Fax Volume		
Jitter Value (Fax)		
Mode (Fax)		

Table is continued on next page…

Voice/Fax Parameter Definitions (continued)		
Coder Parameters		
Coder	Manual or Automatic	Determines whether selection of coder is manual or automatic. When Automatic is selected, the local and remote voice channels will negotiate the voice coder to be used by selecting the highest bandwidth coder supported by both sides without exceeding the Max Bandwidth setting. G.723, G.729, or G.711 are negotiated.
Selected Coder	**G.711** a/u law 64 kbps; **G.726**, @ 16/24/32/40 kbps; **G.727**, @ nine bps rates; **G.723.1** @ 5.3 kbps, 6.3 kbps; **G.729**, 8kbps; **Net Coder** @ 6.4, 7.2, 8, 8.8, 9.6 kbps	This is applicable to outgoing IP calls only. Select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling must have the same voice coder selected. Default = G.723.1 @ 6.3 kbps. Here 64K of digital voice is compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one. To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Selected Coder additional choices	**G.711, G.729** -or- **G.729, G.711**	Coder Priority has two options (G.711, G.729 or G.729, G711) on the Selected Coder listing of the Coder group on the Voice/Fax screen. If G.711 is the higher priority, i.e., G.711 is preferred to G729 on the sending side, then G.711, G.729 option is selected. Similarly, if G.729 has the higher priority, then G.729, G.711 option is selected. It is used whenever a user wants to advertise both G.711 and G.729 coders with higher preference to a particular coder. It is useful when the calls are made from a particular channel on the VOIP to two different destinations where one supports G.711 and the other supports G.729.
Max bandwidth (coder)	11 – 128 kbps	This drop-down list enables you to select the maximum bandwidth allowed for this channel. The Max Bandwidth drop-down list is enabled only if the Coder is set to Automatic. If coder is to be selected automatically ("Auto" setting), then enter a value for maximum bandwidth.
Advanced Features		
Silence Compression	checkbox	Determines whether silence compression is enabled (checked) for this voice channel. With **Silence Compression** enabled, the MultiVOIP GSM will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel (*default = on*).
Echo Cancellation	checkbox	Determines whether echo cancellation is enabled (checked) for this voice channel. **Echo Cancellation** removes echo and improves sound quality (*default = on*).
Forward Error Correction	checkbox	Determines whether forward error correction is enabled (checked) for this voice channel. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC doubles the overhead to the total network bandwidth consumed by the voice channel (*default = Off*).

Table is continued on next page…

Voice/Fax Parameter Definitions (continued)		
Field Name	**Values**	**Description**
AutoCall Parameters		
Auto Call	None, AutoCall	The **AutoCall** option enables the local MVPGSM to call a remote SIP endpoint without the user having to dial a Phone Directory Database number. As soon as you access the channel, the MVPGSM immediately connects to the SIP end point identified in the **Phone Number** box of this option. This function requires an entry in the Outbound Phone Book of the local MVPGSM.
Generate Local Dial Tone	Y/N	If selected, dial tone will be generated locally while the call is connecting between gateways. This would be useful if there is a long network delay.
Phone Number	--	Phone number used for Auto Call function. This phone number must correspond to an entry in the Outbound Phone Book of the local MVPGSM.
Dynamic Jitter		
Dynamic Jitter Buffer		**Dynamic Jitter** defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from an SIP end point, varying delays between packets may occur due to network traffic problems. This is called Jitter. To compensate, the MVPGSM uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MVPGSM to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An Optimization Factor adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly affects the voice delay between MultiVOIP GSM gateways.
Minimum Jitter Value	60 - 400ms	The minimum dynamic jitter buffer of 60 milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 60 ms
Maximum Jitter Value	60 - 400ms	The maximum dynamic jitter buffer of 400 milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 ms
Optimization Factor	0 - 12	The **Optimization Factor** determines how quickly the length of the Dynamic Jitter Buffer is changed based on actual jitter encountered on the network. Selecting the minimum value of 0 means low voice delay is desired, but increases the possibility of jitter-induced voice quality problems. Selecting the maximum value of 12 means highest voice quality under jitter conditions is desired at the cost of increased voice delay. Default = **7**.
Auto Disconnect		
Automatic Disconnection	--	The **Automatic Disconnection** group provides four options which can be used singly or in any combination.
Jitter Value	1-65535	The **Jitter Value** defines the average inter-arrival packet deviation before the call is automatically disconnected. The default is **300** milliseconds. A higher value means voice transmission will be more accepting of jitter. A lower value is less tolerant of jitter. Inactive by default. When active, default = 300 ms. However, value must equal or exceed Dynamic Minimum Jitter Value.
Call Duration	1-65535	**Call Duration** defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. Inactive by default. When active, default = **180** sec.
Consecutive Packets Lost	1-65535	**Consecutive Packets Lost** defines the number of consecutive packets that are lost after which the call is automatically disconnected. Inactive by default. When active, default = **30**
Network Disconnection	1 - 65535	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost. Default = 30 sec

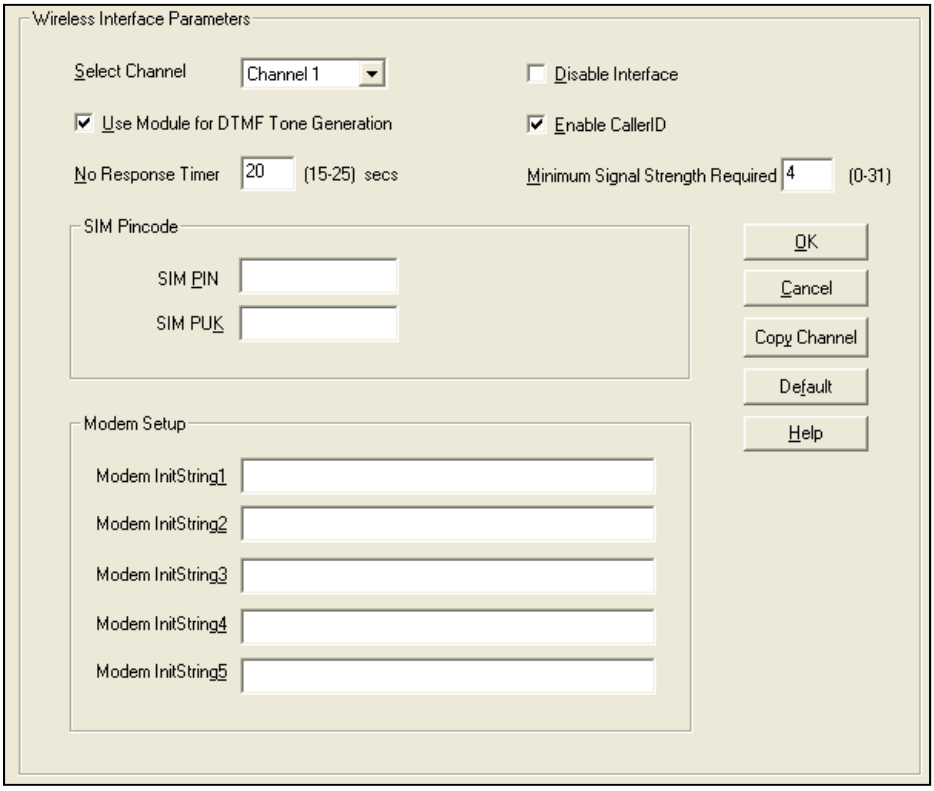
Configurable Payload Type

The Configurable Payload Type is located on the bottom of the Voice/Fax screen. The Configurable Payload Type is used when the remote side uses a different payload type for the associated features.

Wireless Setup

Wireless Interface Parameters

The Wireless Interface parameters are set individually for each channel. In each field, enter the values that fit your particular setup. Once you have established a set of Interface parameters for a channel, you can apply that entire set of parameters to another channel by using the Copy Channel button.



Wireless Interface Parameter definitions		
Field Name	**Values**	**Description**
Select Channel	number	Select which channel you want to set parameters for.
Disable Interface	checkbox	Enable or Disable the interface.
Use Module for DTMF Generation	checkbox	Enable or Disable DTMF generation. When checked, the wireless module will generate DTMF tones to the cell network. If this is enabled, out of band DTMF must be set in Voice/Fax and the matching DTMF out of band method must be set on the remote SIP end point as well (RFC2833 or SIP info method). When unchecked (Disabled), the DSP will generate DTMF tones to the cell network.
Enable Caller ID	checkbox	Enable or Disable Caller ID reporting. This allows Caller ID from the cell network to be passed over IP.
No Response Timer	15 - 25	Set the timeout in seconds, for the wireless module to wait for a valid response from AT commands.
Minimum Signal Strength Required	0 - 31	Set the lowest allowed signal quality for the wireless channel to initiate a call.
SIM Pincode		
SIM PIN	numbers	Should your SIM card get locked out, your provider can give you the PIN code for the SIM on this channel.
SIM PUK	numbers	Should your SIM card get locked out, your provider can give you the PUK code for the SIM on this channel.
Modem Setup	Five areas are available for initialization strings to be sent to the internal wireless modules.	
Copy Channel	(button)	Will copy all settings from the current channel to the other available channel.
Default	(button)	Resets parameters back to their factory default settings.

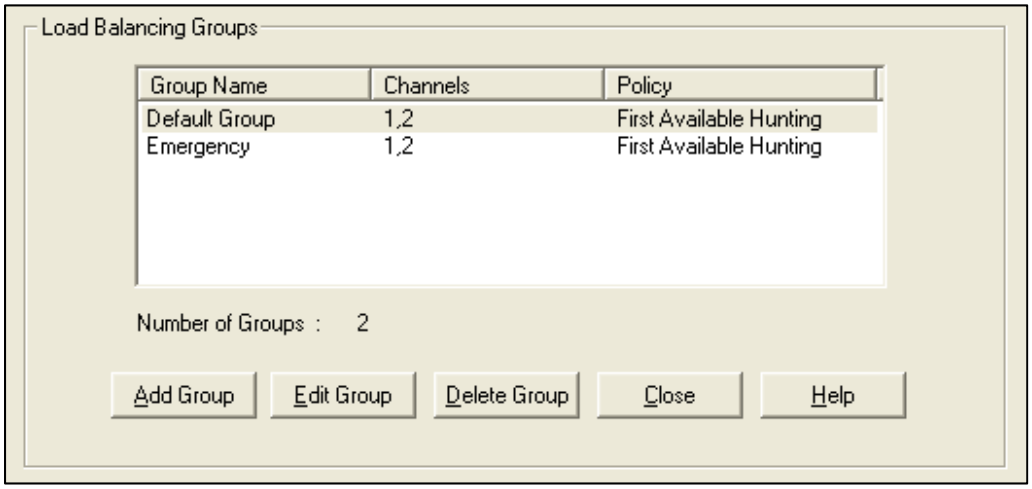
Load Balancing

There are three types of Load Balancing available for calls originating from the IP network and going out to the cellular network: First Available Hunting (often just referred to as 'Hunting'), Least Used First (abbreviated as LUF), and Round Robin (abbreviated as RR).

First Available Hunting:
First Available Hunting will search through the available channels and use the first SIM that presents itself as available for use.

Least Used First:
Least Used first will compare the calling plans for each channel and select the one that has the most remaining minutes available. This is determined by dividing the Minutes remaining / Days remaining and the channel with the largest result remaining will be used first.

Round Robin:
With Round Robin load balancing, the channels are treated as a list and once a channel is used, it goes to the bottom of the list and the next channel comes to the top of the list and is ready for use.



Load Balancing parameters		
Field Name	**Values**	**Description**
Group Name	Text	A descriptive name given to the created group.
Channels	number(s)	Lists the channels that are using the load balance group.
Policy	Text	Displays which load balancing type is in use for the given group.
Number of Groups	number	Displays the total number of load balancing groups that have been created.
Add Group	(button)	Clicking this button will bring up the Add/Edit Group screen allowing you to create a new group or change the settings of an existing group.
Edit Group	(button)	Clicking this button will bring up the Add/Edit Group screen allowing you to create a new group or change the settings of an existing group.
Delete Group	(button)	This will permanently delete the selected group. Please verify that the correct group is highlighted in the Load Balancing Groups window before clicking the Delete Group button.
Close	(button)	This will close the Load Balancing Groups screen.

Load Balancing continued below…

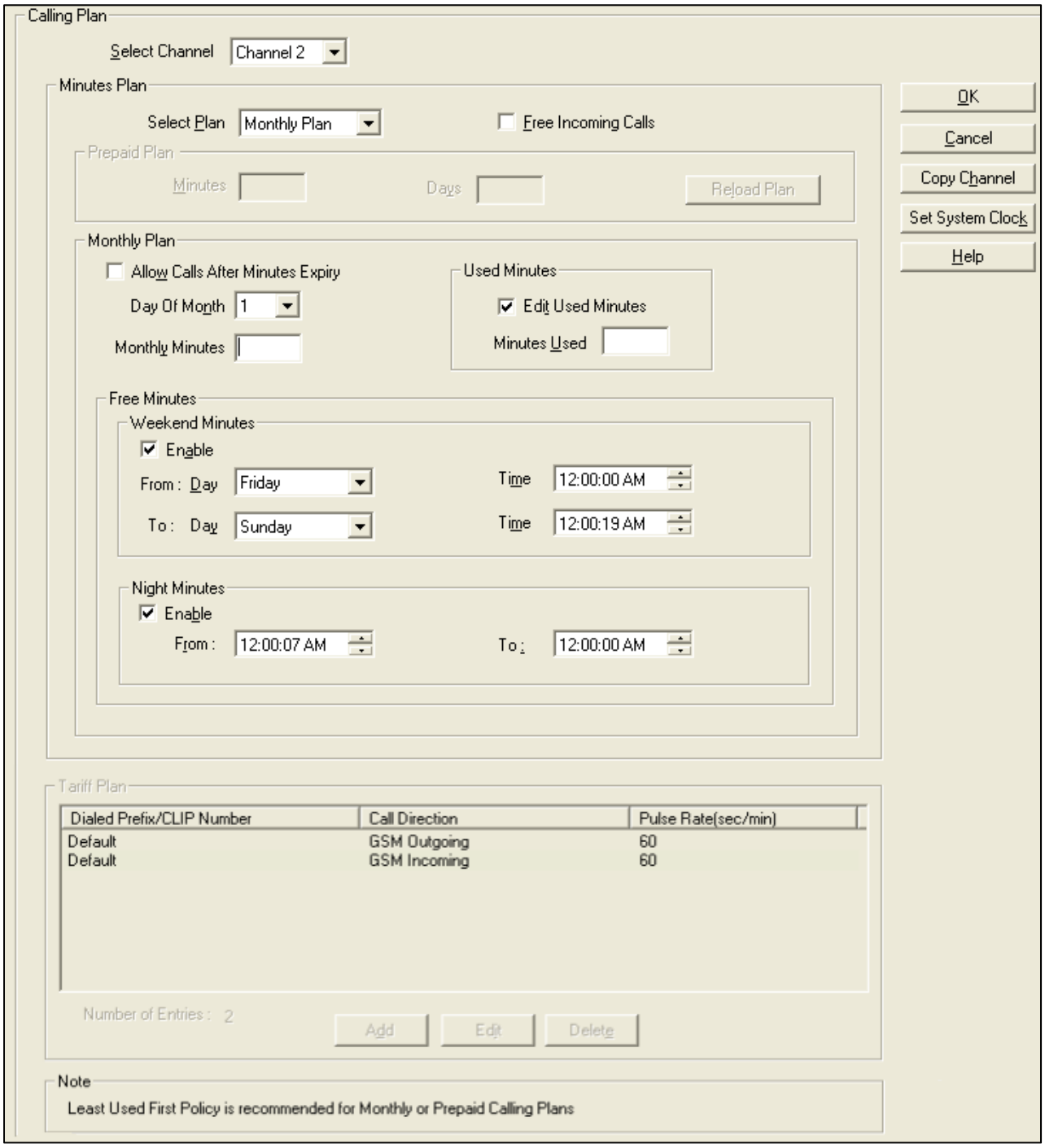
Add/Edit Group

Selecting either the Add Group or Edit Group will bring you to the Add/Edit Group screen where you can name the group you are creating, select the type of load balancing to be used and what channels it will apply to.

Add/Edit Group parameters		
Field Name	**Values**	**Description**
Group Name	alpha numeric	Enter a name for this Load Balancing group that will help you remember it.
Policy	"First Available Hunting," "Least Used First," or "Round Robin"	Select one of the three policies, as appropriate for how you want the channel(s) to be used. • First Available Hunting will search through the available channels and use the first SIM that presents itself as available for use. • Least Used first will compare the calling plans for each channel and select the one that has the most remaining minutes available. This is determined by dividing the Minutes remaining / Days remaining and the channel with the largest result remaining will be used first. • With Round Robin load balancing, the channels are treated as a list and once a channel is used, it goes to the bottom of the list and the next channel comes to the top of the list and is ready for use.
Select Channels	(check boxes)	Check or un-check the channel boxes to have this load balancing group applied only to the needed channels.

Calling Plan

When the Least Used First load balancing is selected for a channel or the SIM has a monthly or prepaid allowance, the Calling Plan screen is used to determine the number of minutes and number of days remaining on the channel/SIM. The Calling Plan screen defaults to 'Unlimited' for every channel.



Calling Plan continued:

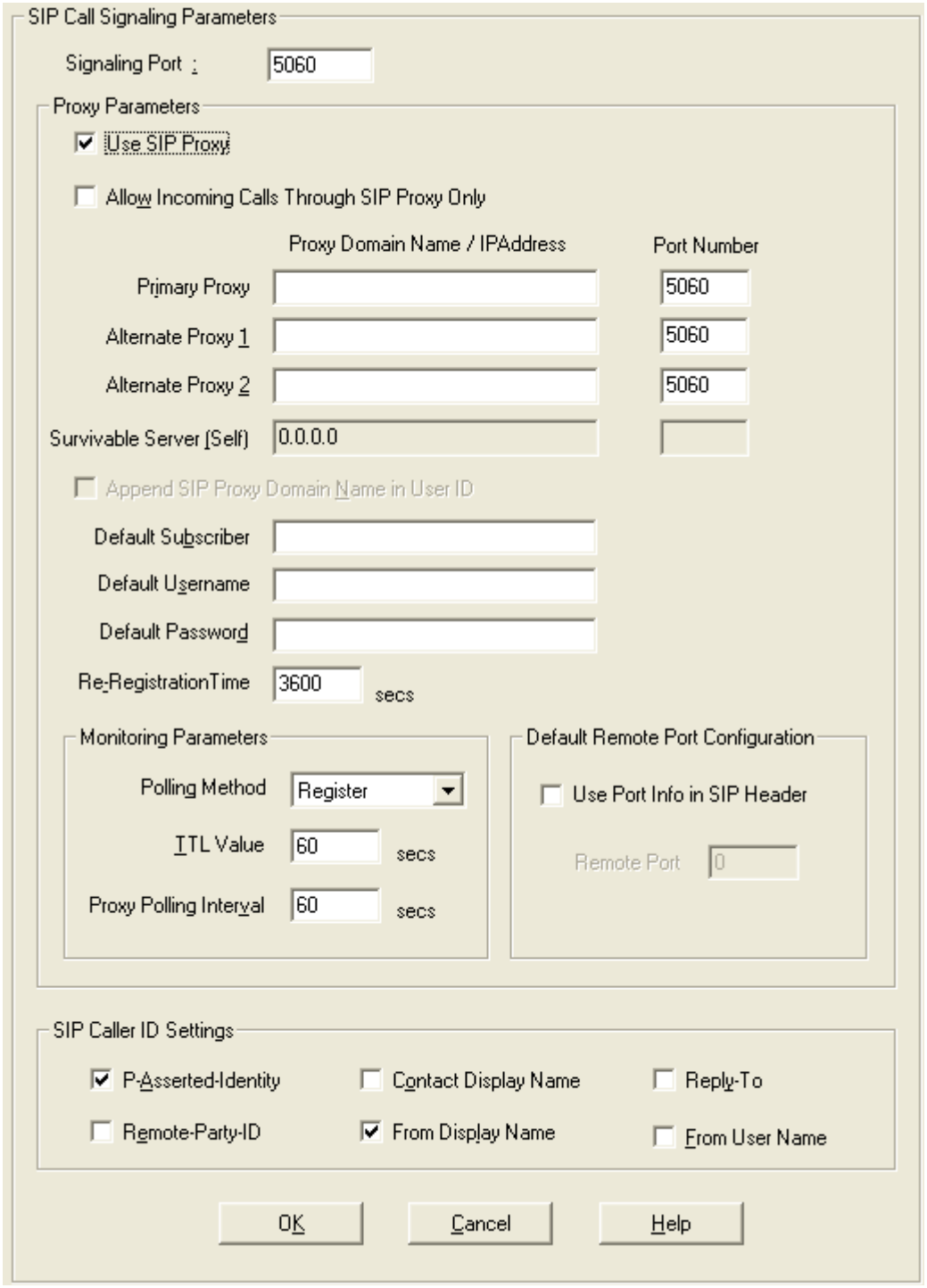
Calling Plan parameters		
Field Name	**Values**	**Description**
Minutes Plan section		
Select Plan	Unlimited, Monthly Plan, Prepaid Plan	For each channel, you can select a Monthly or Prepaid plan that will be used for the channel, or alternatively, you can select 'Unlimited' to leave the Prepaid and Monthly plan sections grayed-out as they are not needed. **Note:** The Calling Plan screen is only needed with Least Used First load balancing.
Free Incoming Calls	(check box)	Check this box if incoming calls are not charged to the calling plan.
Prepaid Plan section		
Minutes	number	Enter the total number of Prepaid Minutes available to this channel/SIM.
Days	number	Enter the total number of Prepaid Days available to this channel/SIM.
Reload Plan	(button)	Click the Reload Plan button to use the current prepaid plan again.
Monthly Plan section		
Allow Calls After Minutes Expiry	(check box)	Check this box if you want the MVPGSM to allow calls to be routed through this channel after its minutes have expired for the current term.
Monthly Minutes	number	Enter the number of minutes available each month for this channel/SIM.
Day of Month	number	Enter the recharge day of the month for this channel (SIM).
Edit Used Minutes	number	If you want to override the calculated 'Minutes Used' field, check this box and the Minutes Used will be set to '0' and you can enter a new value.
Used Minutes	number	Each time this page is accessed, this will display the minutes used so far on this channel (SIM).
Free Minutes section		
Weekend Minutes:		
Enable	(check box)	Check this box if free weekend times are part of this plan.
From:	Day & Time	Enter the Day and Time when the free weekend minutes start.
To:	Day & Time	Enter the Day and Time when the free weekend minutes stop.
Night Minutes:		
Enable	(check box)	Check this box if there are cost differences for night times.
From:	Time	Enter the Time when the free night minutes start.
To:	Time	Enter the Time when the free night minutes stop.
Tariff Plan section		
Dialed Prefix/CLIP Number	Name or number	The name (there are two "default" plans already in the system) or number associated with a tariff plan.
Call Direction	Outgoing or Incoming	This displays the direction that is associated with the plan (either GSM Incoming or GSM Outgoing).
Pulse Rate	sec/min	Displays the number of seconds that need to pass per minute of call time deducted.
Number of Entries	number	This displays the total number of Tariff Plan entries for this channel.
Add	(button)	This will open the Add/Edit Tariff Plan screen.
Edit	(button)	This will open the Add/Edit Tariff Plan screen.
Delete	(button)	This will permanently delete the selected plan.

Call Signaling

The MultiVOIP GSM uses SIP call signaling.

SIP

Session Initiation Protocol is the only option available for application layer control of the MultiVOIP GSM. The fields are detailed in the table below (the table spans two pages).



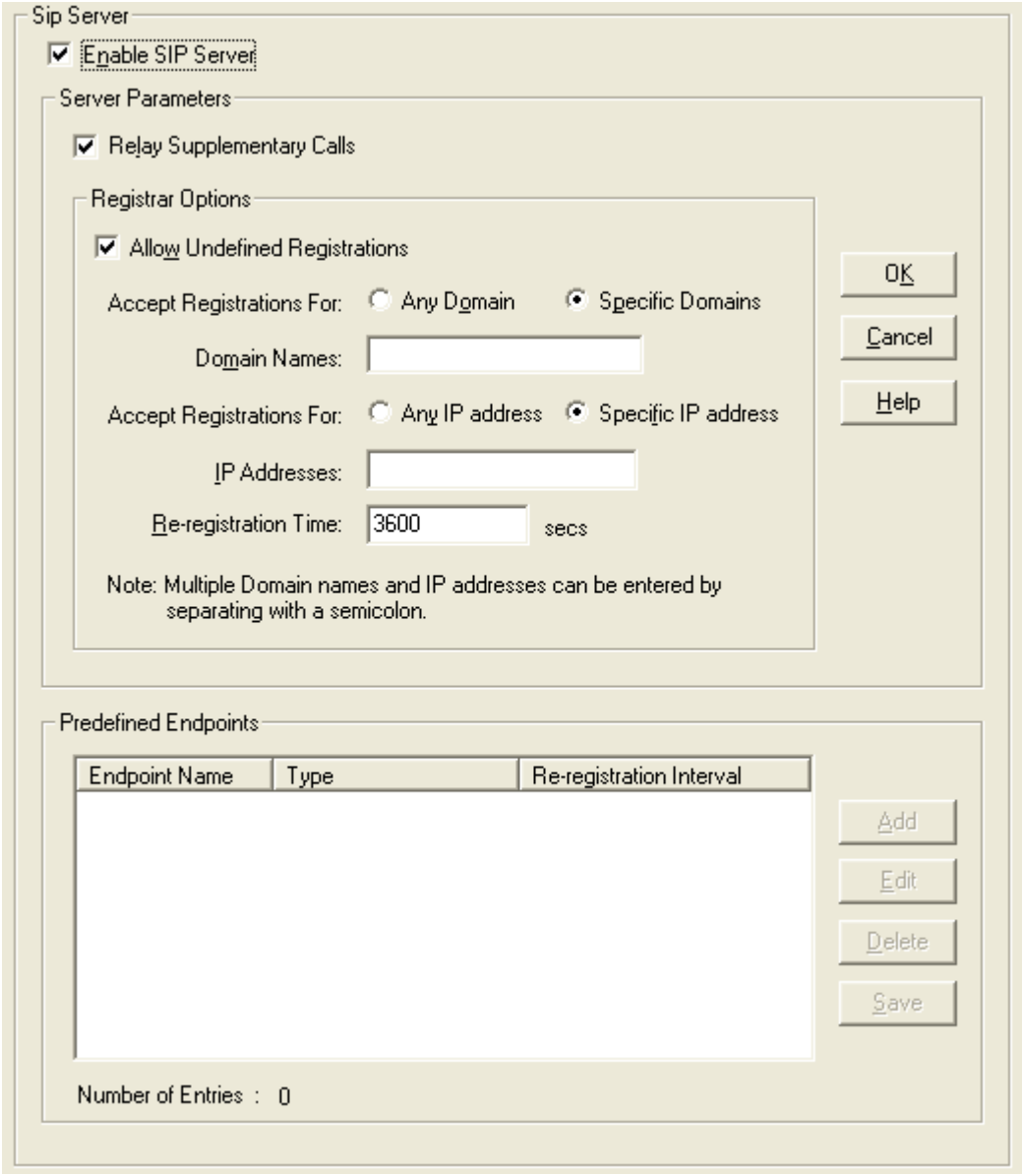
SIP Call Signaling Parameter Definitions		
Field Name	**Values**	**Description**
SIP Parameters		
Signaling Port	*port*	Port number on which the MultiVOIP GSM UserAgent software module will be waiting for any incoming SIP requests. Default = 5060
Use SIP Proxy	checkbox	Allows the MultiVOIP GSM to work in conjunction with a proxy server.
Allow Incoming Calls Through SIP Proxy Only	checkbox	When selected, incoming calls are accepted only if those calls come through the proxy.
Proxy Parameters		
Primary Proxy	*n.n.n.n*	Network address of the preferred SIP proxy server for controlling the traffic of the current VOIP.
Alternate Proxy 1 and 2	--	A first and a second alternate SIP proxy server can be specified for use by the VOIP for situations where the Primary proxy server is otherwise unavailable.
Survivable Server (Self)	*n.n.n.n*	This will display the IP address of the SIP Survivable server.
Append SIP Proxy Domain Name in User ID	checkbox	When checked, the domain name of the SIP Proxy serving the MultiVOIP GSM gateway will be included as part of the User ID for that gateway. If unchecked, the SIP Proxy's IP address will be included as part of the User ID instead of the SIP Proxy's domain name.
Port Number	*port*	Logical port number for proxy communications. Default = 5060
Default Subscriber		This is used as the default end point register with a Proxy.
Default Username	*name*	If the Username is not populated in the Phone Book, this is the Username that will be used. This works the same for the password as well.
Password	*password*	Password for proxy server function. See "Default Username" description above.
Re-Registration Time	10–65535 seconds	This is the timeout interval for registration of the MultiVOIP GSM with a SIP proxy server. The time interval begins the moment the MultiVOIP GSM gateway registers with the SIP proxy server and ends at the time specified by the user in the Re-Registration Time field (this field). When/if registration lapses, call traffic routed to/from the MultiVOIP GSM through the SIP proxy server will cease. However, calls in progress will continue to function until they end.
Monitoring Parameters		
Polling Method	Register -or- Options	
TTL Value	SIP proxy "Time to Live" value. (*in seconds*)	As soon as a MultiVOIP GSM gateway registers with a SIP proxy server (allowing the proxy server to control its call traffic) a countdown timer begins. The TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP GSM gateway needs to reconfirm the availability of the proxy server with which it is currently registered to. This is required, as typically the re-registration time will be on an hourly basis. Whereas to reduce the call failure rates, periodically (based on the configured TTL value), the MVPGSM has to confirm the availability of the proxy server with which it is registered to. If the proxy server is not available, the MultiVOIP GSM gateway will switch over to the next level of the proxy server.
Proxy Polling Interval	60 - 300	The interval between the VOIP SIP server's successive attempts to connect to and be governed by a higher level SIP proxy server. The Primary Proxy is the highest level SIP server. Alternate Proxy 1 is second; Alternate Proxy 2 is the lowest order SIP proxy server.

SIP Call Signaling Parameters continued below…

Default Remote Port Configuration		
Use Port Info in SIP Header	checkbox	Check this box if you want to include the port information in the SIP header. This will allow the box below to be populated with the remote port number.
Remote Port	*port*	Enter the remote port number that is used.
SIP Caller ID Settings		
P-Asserted Identity	checkbox	Check this box if you want to send the CLIP information using the P-Asserted Identity header. Example CLIP number: "**+15551234567**" P-Asserted Identity: "+15551234567"<sip:+15551234567@192.168.54.23>;user=phone
Contact Display Name	checkbox	Check this box if you want to send the CLIP information using the Contact header. Example CLIP number: "**+15551234567**" *101 – Registered Number with proxy for that channel* Contact: "+15551234567"<sip: 101@192.168.53.176>
Reply-To	checkbox	Check this box if you want to send the CLIP information using the Reply-To header. Example CLIP number: "**+15551234567**" Reply-To: "+15551234567"<sip: +15551234567@192.168.53.23>
Remote-Party-ID	checkbox	Check this box if you want to send the CLIP information using the Remote-Party-ID header. Example CLIP number: "**+15551234567**" Remote-Party-ID: "+15551234567"<sip: +15551234567@192.168.53.23>; party=calling; id-type=subscriber; privacy=off; screen=yes
From Display Name	checkbox	Check this box if you want to send the CLIP information using the Display name portion of the 'From' header. 'From' header syntax is: DisplayName <Username@IP address> Example CLIP number: "**+15551234567**" From: "+15551324567"<sip:+15551234567@192.168.53.176>;user=phone
From User Name	checkbox	Check this box if you want to send the CLIP information using the Username name portion of the 'From' header. 'From' header syntax is: DisplayName <Username@IP address> Example CLIP number: "**+15551234567**" From: "+15551324567"<sip:+15551234567@192.168.53.176>;user=phone

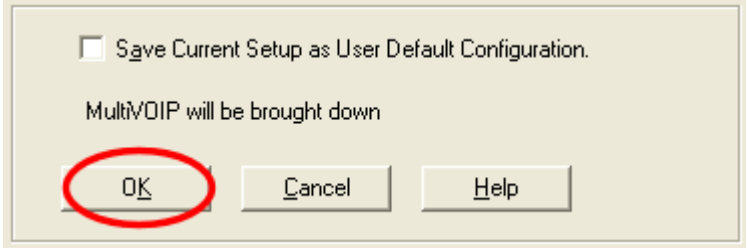
Sip Server

Enter the information for the SIP server here.



Descriptions for the various settings of the SIP Server screen can be found in the table below.

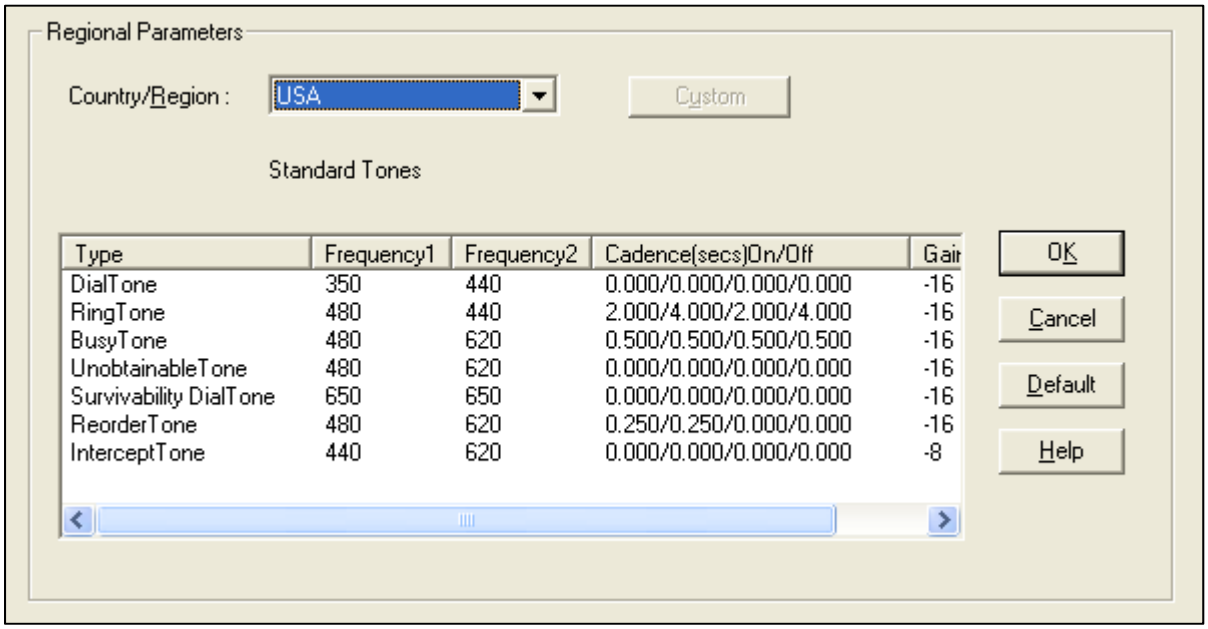
Important Note: After entering the information for a new endpoint, you will need to click the OK button. This will then open the *Help: About* screen. You now must go to the **Save Setup: Save & Reboot** screen and click the **OK** button here to save what you have entered and allow the MVPGSM to restart. If you do not do this, your endpoint will not be saved to the unit and the section will remain blank.



SIP Server Definitions		
Enable SIP Server	checkbox	Check this box to enable the MVPGSM to work with a SIP server.
Server Parameters		
Relay Supplementary Calls	checkbox	If you want all supplementary calls to be relayed by the SIP server, check this box.
Registrar Options		
Allow Undefined Registrations	checkbox	Check this box if you want to allow undefined registrations.
Accept Registrations For: [Domain]	selection	Select to either allow 'Any Domain' or 'Specific Domains' to manually enter domains that will be allowed for registrations.
Domain Names	name	If you selected 'Specific Domains' above, you need to enter the domains here.
Accept Registrations For: [IP Address]	selection	Select to either allow 'Any IP Address' or 'Specific IP Addresses' to manually enter addresses that will be allowed for registrations.
IP Addresses	*n.n.n.n*	If you selected 'Specific IP Addresses' above, you need to enter the addresses here.
Re-registration Time	value	Enter the time to wait between a registration attempts. Value is in seconds. Default is 3600 seconds. Range allowed is 10- 65535 seconds. Expiration of the registration interval means that the gateway has lost contact with the main SIP server and that the unit will enter its 'survivability' mode. In survivability mode, the unit will complete calls acting as a backup to the main SIP server. Normally, however, the MVPGSM will initiate re-registration with some small margin of time before the interval lapses.
Predefined Endpoints		
Endpoint Name	name	This column will display a list of endpoint names.
Type	type	This column will display the endpoint types associated with the names.
Re-registration Interval	time (in seconds)	This column will display the re-registration interval that is set for each endpoint.
Add	button	Click this button to add a predefined endpoint to the list.
Edit	button	Click this button to edit a predefined endpoint in the list.
Delete	button	Click this button to delete a predefined endpoint from the list.
Save	button	Click this button to save your changes.

Regional

The Regional Parameters are used to set the phone signaling tones and cadences. For the country selected, the standard set of frequency pairs will be listed for dial tone, busy tone, 'unobtainable' tone (fast busy or trunk busy), ring tone, and other, more specialized tones. If you need settings that are not available, the Custom selection will let you set the tones to what is necessary. The Regional Parameters fields are described in the table below.



"Regional Parameter" Definitions		
Field Name	**Values**	**Description**
Country/Region	USA, Japan, UK, Custom	Name of a country or region that uses a certain set of tone pairs for **dial tone, ring tone, busy tone**, **unobtainable tone** (fast busy tone), **survivability tone** (tone heard briefly, 2 seconds, after going off hook denoting survivable mode of VOIP unit), **re-order tone** (a tone pattern indicating the need for the user to hang up the phone), and **intercept tone** (a tone that warns a party that has gone off hook but has not begun dialing, within a prescribed time, that an automatic emergency or attendant number will be called; the automatic call can be used to direct an attendant's attention to a disabled or distressed caller, allowing an appropriate response to be made). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The "Custom" option (button) assures that any tone-pairing scheme worldwide can be accommodated.
Standard Tones fields		
Type column	dial tone, ring tone, busy tone, unobtainable tone, survivability tone, re-order tone	Type of telephony tone-pair for which frequency, gain, and cadence are being presented.
Frequency 1	freq. in Hertz	Lower frequency of pair.
Frequency 2	freq. in Hertz	Higher frequency of pair.
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This applies to the dial, ring, busy and 'unobtainable' tones that the MultiVOIP GSM outputs as audio to the wireless port. **Default: -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This applies to the dial, ring, busy, and 'unobtainable' (fast busy) tones that the MultiVOIP GSM outputs as audio to the wireless port. **Default: -16dB**
Cadence (ms) On/Off	n/n/n/n four integer time values in milliseconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable (fast busy), dial tone ("0" indicates continuous tone), survivability, and re-order. Default values differ for different countries/regions. Although most cadences have only two parts (an "on" duration and an "off" duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.
Custom (button)	Only available when Country/Region is set to "Custom"	Click on the "Custom" button to bring up the **Custom Tone Pair Settings** screen. This screen allows the user to specify tone pair attributes that are not found in any of the standard national/regional telephony toning schemes.

Setting Custom Tones and Cadences (optional). The Regional Parameters dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tones, dial-tones, busy-tones or "unobtainable" tones or "re-order" tones or "survivability" tones for your system. This screen allows the user to specify tone-pair attributes that are not found in any of the standard national/regional telephony toning schemes. To access this customization feature, click on the **Custom** button on the **Regional Parameters** screen. The "Custom" button is active only when "Custom" is selected in the **Country/Region** field.

Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Tone Pair	dial tone, busy tone ring tone, 'unobtainable' tone, survivability tone, re-order tone	Identifies the type of telephony signaling tone for which frequencies are being specified.
Tone Pair Values		**About Defaults**: US telephony values are used as defaults on this screen.
Frequency 1	Frequency in Hertz	Lower frequency of pair.
Frequency 2	Frequency in Hertz	Higher frequency of pair.
Gain 1	+3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This figure describes amplification that the MultiVOIP GSM applies to outbound tones entering the MultiVOIP GSM at the input port. **Default: -16dB**
Gain 2	+3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP GSM applies to outbound tones entering the MultiVOIP GSM at the input port. **Default: -16dB**
Cadence 1	integer time value in milliseconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, dial tone ("0" indicates continuous tone) survivability and re-order. Cadence 1 is duration of first period of tone being "on" in the cadence of the telephony signal.
Cadence 2	duration in milliseconds	Cadence 2 is duration of first "off" period in signaling cadence.
Cadence 3	duration in milliseconds	Cadence 3 is duration of second "on" period in signaling cadence.
Cadence 4	duration in milliseconds	Cadence 4 is duration of second "off" period in the signaling cadence.

SMTP

Setting the SMTP Parameters (Log Reports by Email). The SMTP Parameters screen is applicable when the VOIP administrator has chosen to receive log reports by email (this is done by selecting the "SMTP" checkbox in the Others screen and selecting "Enable SMTP" in the SMTP Parameters screen.)

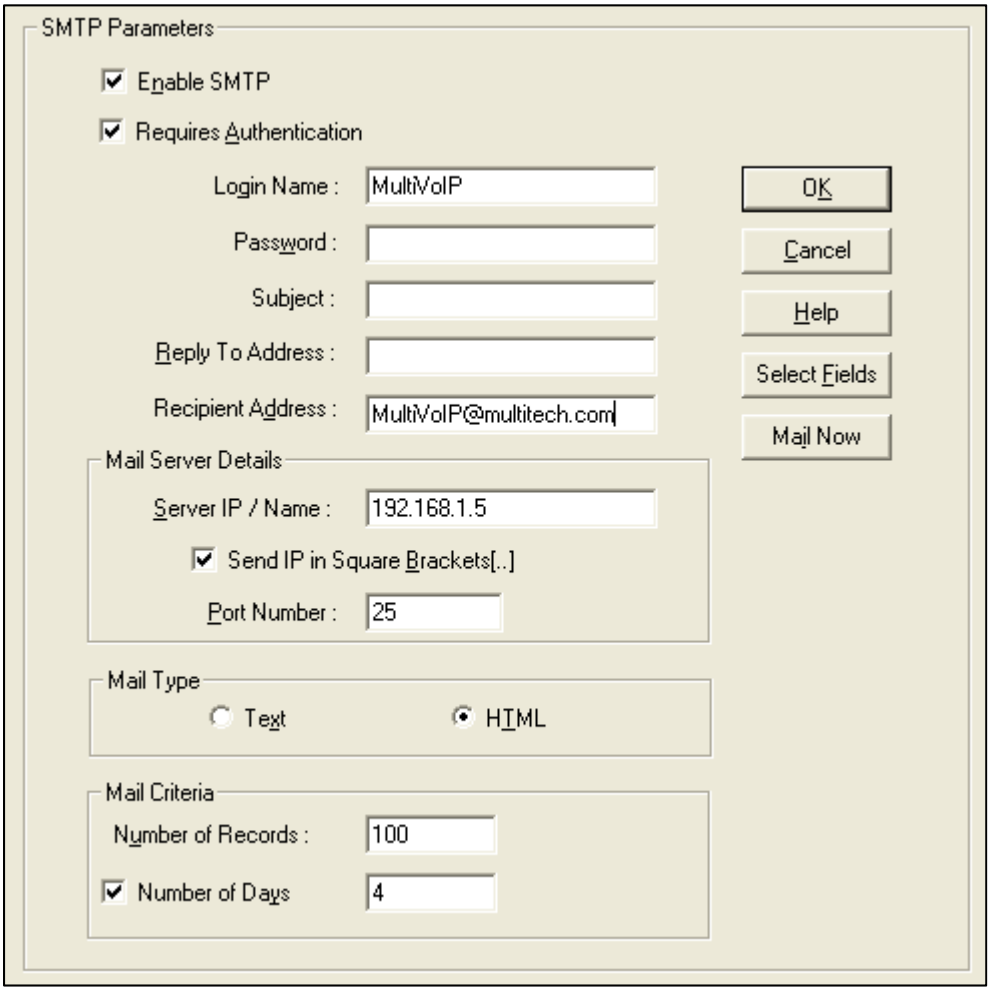
Email Address for VOIP (for email call log reporting)

This is needed only if log reports of VOIP call traffic are to be sent by email.

Ask Mail Server administrator to set up email account (with password) for the MultiVOIP GSM unit itself. Be sure to give a unique identifier to each individual MultiVOIP GSM unit. Get the IP address of the mail server computer, as well.

MultiVOIP GSM as Email Sender. When SMTP is used, the MultiVOIP GSM will actually be given its own email account (with Login Name and Password) on some mail server connected to the IP network. Using this account, the MultiVOIP GSM will then send out email messages containing log report information. The "Recipient" of the log report email is ordinarily the VOIP administrator. Because the MultiVOIP GSM cannot receive email, a "Reply-To" address must also be set up. Ordinarily, the "Reply-To" address is that of a technician who has access to the mail server or MultiVOIP GSM or both, and the VOIP administrator might also be designated as the "Reply-To" party. The main function of the Reply-To address is to receive error or failure messages regarding the emailed reports.

The **SMTP Parameters** screen is shown below:



"SMTP Parameters" Definitions		
Field Name	**Values**	**Description**
Enable SMTP	checkbox	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select "SMTP" in the **Logs** screen.
Requires Authentication	checkbox	If this is checked, the MultiVOIP GSM will send Authentication information to the SMTP server. The authentication information indicates whether or not the email sender has permission to use the SMTP server.
Login Name	alpha-numeric	This is the User Name for the MultiVOIP GSM unit's email account.
Password	alpha-numeric	Login password for MultiVOIP GSM unit's email account.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP GSM unit.
Reply-To Address	email address	User specified. This email address functions as a source email identifier for the MultiVOIP GSM, which, of course, cannot usefully receive email messages. The Reply-To address provides a destination for returned messages indicating the status of messages sent by the MultiVOIP GSM (esp. to indicate when log report email was undeliverable or when an error has occurred).
Recipient Address	email address	Email address where log reports are sent.
Mail Server Details		
Server IP / Name	*n.n.n.n* or name	This is the mail server's IP address or resolvable name. This mail server must be accessible on the IP network to which the MultiVOIP GSM is connected.
Send IP in Square Brackets	checkbox	Check this if you want the IP address sent incased in square brackets "[*n.n.n.n*]".
Port Number	number	The default value, 25, is a standard port number for SMTP.
Mail Type	text or html	Mail type in which log reports will be sent.
Mail Criteria		Criteria for sending log summary by email. The log summary email will be sent out either when the user-specified number of log messages has accumulated, or once every day or multiple days, *whichever comes first*.
Number of Records	number	This is the number of log records that must accumulate to trigger the sending of a log-summary email.
Number of Days	number	This is the number of days that must pass before triggering the sending of a log-summary email.

The **SMTP Parameters** dialog box has a secondary dialog box, accessed by the *Select Fields* button, which allows you to customize email logging. The MultiVOIP GSM software logs data about many aspects of the call traffic going through the MultiVOIP GSM. The Custom Fields screen lets you pick which aspects will be included in the email log reports.

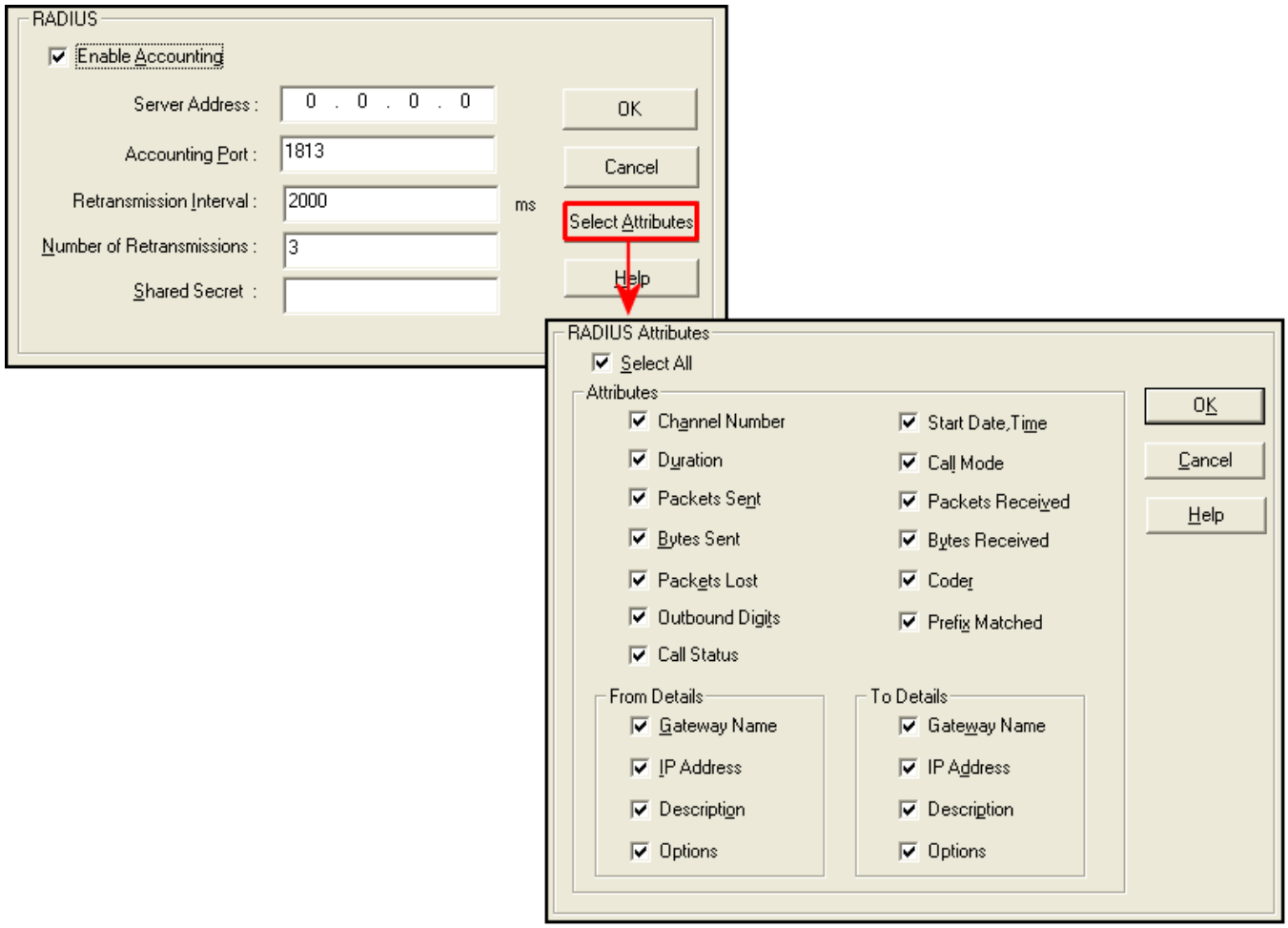
"Custom Fields" Definitions

Field	Description
Select All	Log report to include all fields shown.
Channel Number	Channel carrying call.
Duration	Length of call.
Packets Sent	Total packets sent in call.
Bytes Sent	Total bytes sent in call.
Packets Lost	Packets lost in call.
Outbound Digits Received	The DTMF dialing digits received by this gateway from the remote gateway presuming that DTMF is set to "Out of Band."
Call Status	Successful or unsuccessful.
Call Direction	Indicates originating party.
Server Details	The IP address of the traffic control server (if any) being used will be displayed here if the call is handled through that server.
Disconnect Reason	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause occasioned disconnection (e.g., a technical error or failure). Values are "Normal" and "Local" disconnection.
From Details	
Gateway Name	Originating gateway
IP Address	IP address where call originated.
Description	Identifier of site where call originated.
Options	When selected, log will record Silence Compression and Forward Error Correction by call originator.

Field	Description
Start Date, Time	Date and time the phone call began.
Call Mode	Voice or fax.
Packets Received	Total packets received in call.
Bytes Received	Total bytes received in call.
Coder	Voice Coder /Compression Rate used for call will be listed in log.
Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Type	Indicates the Call Signaling protocol used for the call.
DTMF Capability	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." It can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition.
Outbound Digits Sent	The dialing digits sent by this gateway to the remote gateway presuming that DTMF is set to "Out of Band."
To Details	
Gateway Name	Completing or answering gateway
IP Address	IP address where call was completed or answered.
Description	Identifier of site where call was completed or answered.
Options	When selected, log will record Silence Compression and Forward Error Correction by party answering call.

RADIUS

In general, RADIUS is concerned with authentication, authorization, and accounting. The MultiVOIP GSM supports the accounting and authentication functions. The accounting function is well suited for billing of VOIP telephony services. In the *Select Attributes* secondary screen (accessed by clicking on Select Attributes button), the VOIP administrator can select the parameters to be tallied by the RADIUS server.



The fields of the RADIUS screen are described in the table below.

RADIUS Screen Field Definitions		
Field Name	**Values**	**Description**
Enable Accounting	checkbox	When checked, the MultiVOIP GSM will access the accounting functionality of the RADIUS server.
Server Address	*n.n.n.n*	IP address of the RADIUS server that handles accounting (billing) for the current MultiVOIP GSM unit.
Accounting Port	1 - 65535	IP port number where RADIUS accounting information will be transmitted and received.
Retransmission Interval	50 - 5000 (milliseconds)	If the MultiVOIP GSM sends out a packet to the RADIUS server and doesn't receive a response in the retransmit interval, it will retransmit that packet again and wait the retransmit interval again for a response. How many times it does this is determined by the setting in the **Number of Retransmissions** field.
Number of Retransmissions	0 - 255	
Shared Secret	alpha-numeric	Client encryption key for the current VOIP unit.
Select Attributes	(button)	Gives access to RADIUS Attributes screen. On Attributes screen, one can specify the parameters to be tallied by the RADIUS server for accounting (usually billing) purposes.

The RADIUS dialog box has a secondary dialog box, **RADIUS Attributes**, which allows you to customize accounting information sent to the RADIUS server by the MultiVOIP GSM. The MultiVOIP GSM software logs data about many aspects of the call traffic going through the MultiVOIP GSM. The RADIUS Attributes screen lets you pick which aspects will be included in the accounting reports sent to the RADIUS server.

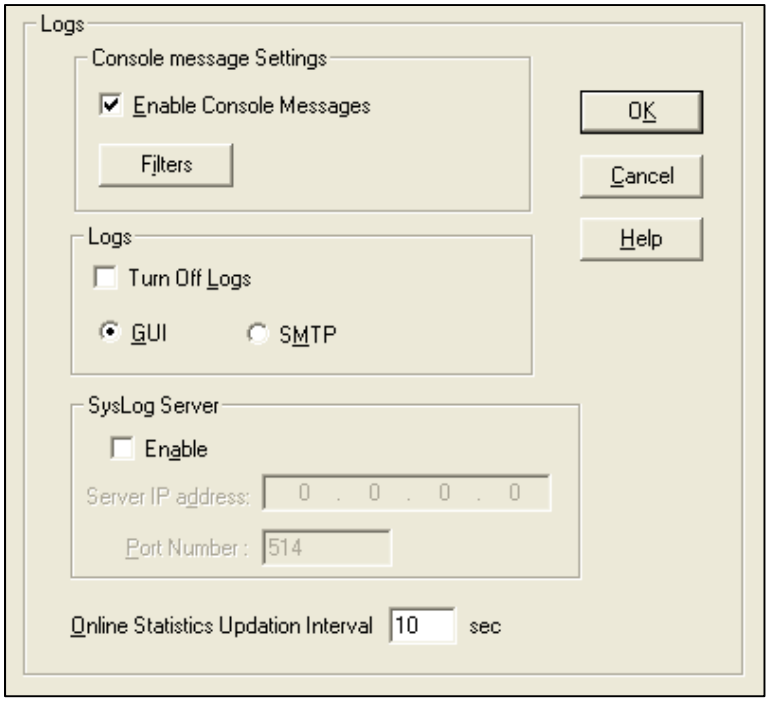
"RADIUS Attributes" Definitions			
Field	**Description**	**Field**	**Description**
Select All	Log report to include all fields shown.		
Channel Number	Channel carrying call.	Start Date, Time	Date and time the phone call began.
Duration	Length of call.	Call Mode	Voice or fax.
Packets Sent	Total packets sent in call.	Packets Received	Total packets received in call.
Bytes Sent	Total bytes sent in call.	Bytes Received	Total bytes received in call.
Packets Lost	Packets lost in call.	Coder	Voice Coder /Compression Rate used for call will be listed in log.
Outbound Digits Sent	DTMF digits received by this gateway from remote gateway (if that DTMF set to "Out of Band").	Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Status	Successful or unsuccessful.		
Server Details	The IP address of the traffic control server being used will be displayed here if the call is handled through that server. The Options field refers to server features that might be activated.		

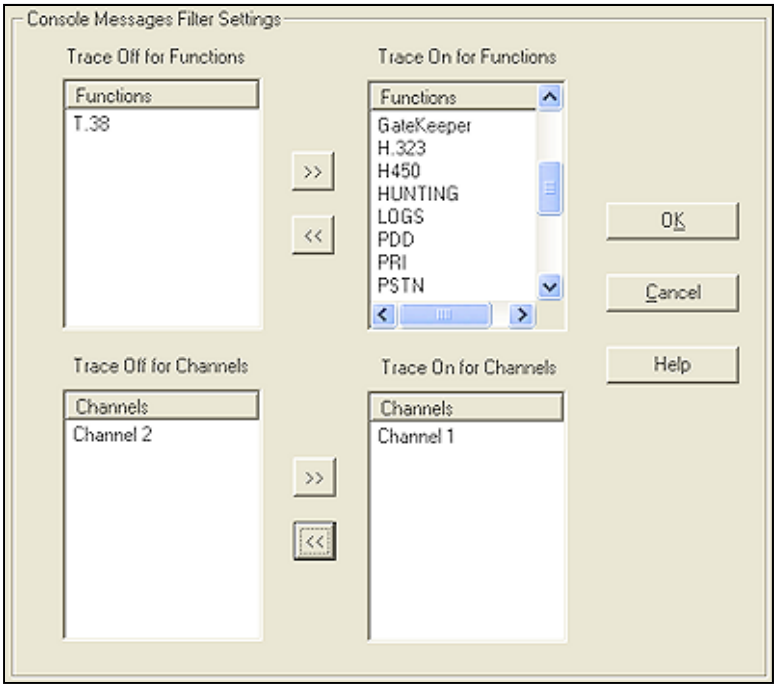
From Details		**To Details**	
Gateway Name	Originating gateway	Gateway Name	Completing or answering gateway
IP Address	IP address where call originated.	IP Address	IP address where call was completed/answered.
Description	Identifier of where call originated.	Description	Identifier of where call was completed/answered.
Options	When selected, log will record Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will record Silence Compression and Forward Error Correction by party answering call.

Logs/Traces

The Logs/Traces screen lets you choose how the VOIP administrator will receive log reports of the console messages over the IP network. Log reports can be received in one of two ways:

- in the MultiVOIP GSM program (interface), or
- via email (SMTP)



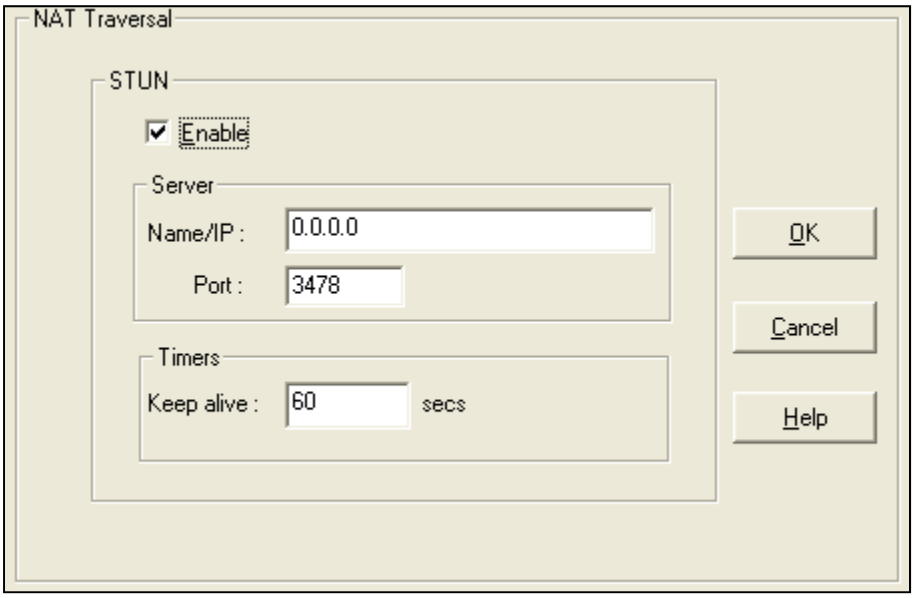


If you enable console messages, you can customize the types of messages to be included/excluded in log reports by clicking on the *Filters* button and using the **Console Messages Filter Settings** screen. If you use the logging function, select the logging option that applies to your VOIP system design. If you intend to use a SysLog Server program for logging, click in that Enable check box. The common SysLog logical port number is 514. If you intend to use the MultiVOIP GSM web browser interface for configuration and control of MultiVOIP GSM units, be aware that the web browser interface does not support logs directly. However, when the web browser interface is used, log files can still be sent to the administrator via email (which requires using SMTP logging).

"Logs" Screen Definitions		
Field Name	**Values**	**Description**
Enable Console Messages	checkbox	Allows MultiVOIP GSM console messages to be read via a basic terminal program like HyperTerminal™ or equivalent. Normally, this should be disabled because it uses MultiVOIP GSM processing resources. Console messages are meant for IT support personnel.
Filters (button)		Click to access secondary screen on where console messages can be included/excluded by category and on a per-channel basis.
Turn Off Logs	checkbox	Check to disable console message logging function.
Logs Buttons		Only one of these log reporting methods, GUI or SMTP, may be chosen.
GUI	•	User must view logs at the MultiVOIP GSM configuration program.
SMTP	•	Log messages will be sent to user-specified email address.
SysLog Server Enable	checkbox	This box must be checked if logging is to be done in conjunction with a SysLog Server program.
IP Address	*n.n.n.n*	IP address of computer, in VOIP network, on which SysLog Server program is running.
Port	number	Logical port for SysLog Server. The default value, 514, is commonly used.
Online Statistics Update Interval	number	Set the interval (in seconds) at which logging information will be updated.

NAT Traversal

Setting the NAT (Network Address Translation) Traversal parameters. STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.

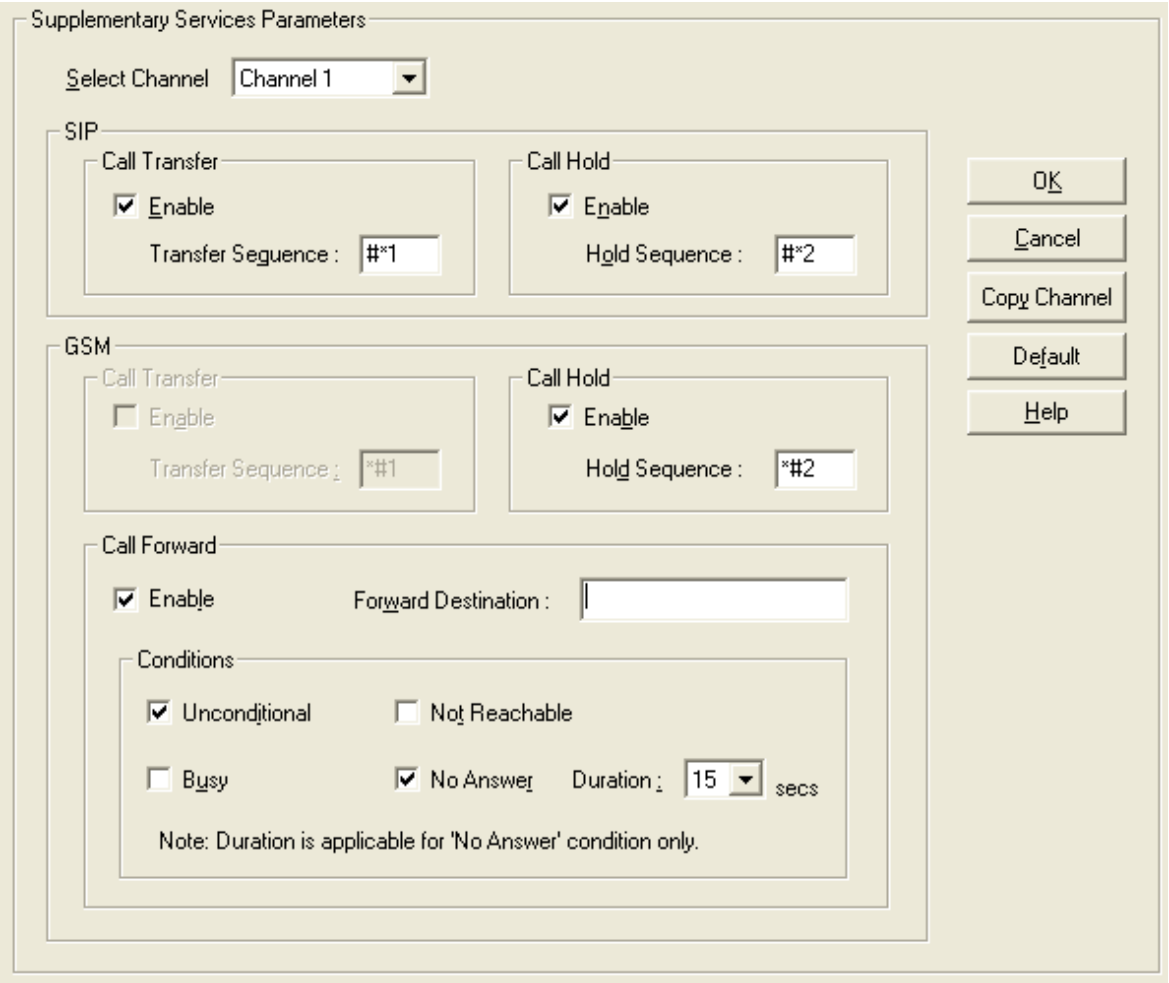


Descriptions for **NAT Traversal** screen fields are presented in the table below.

NAT Traversal Definitions		
Field Name	**Values**	**Description**
Enable (STUN)	checkbox	Enables STUN client functionality in the MultiVOIP GSM. STUN (Simple Traversal of UDP through NATs (Network Address Translation)) is a protocol that allows a server to assist client gateways behind a NAT firewall or router with their packet routing.
Name/IP (Server)	*n.n.n.n*	IP address of the STUN server.
Port (Server; NAT/STUN)	number	The data port (TDM time slot) at which STUN info will be transmitted and received. Default is 3478
Keep Alive (Timers; NAT/STUN)	60 – 3600 (seconds)	"Keep Alive" packets to the STUN server to retain the binding / connection with the STUN server. This is the interval at which the STUN client sends indicator packets to the STUN server to determine whether or not the STUN server is available.

Supplementary Services

Specific supplementary settings can be enabled here on a per-channel basis.



Descriptions of the settings can be found in the table below.

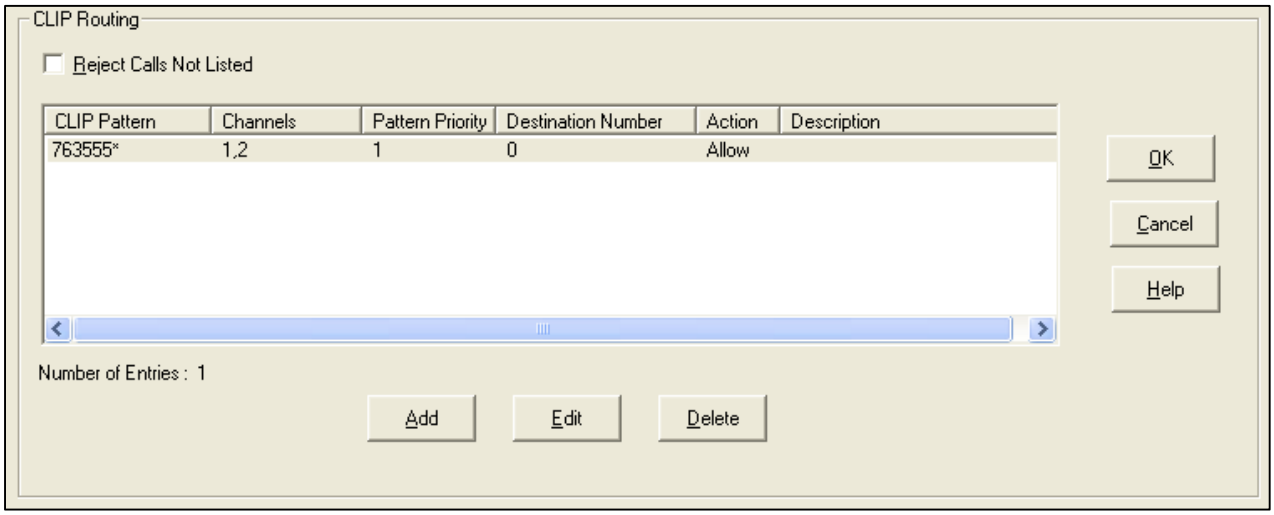
Supplementary Services Parameters		
Select Channel	drop-down	Select the specific channel that you want to apply call services to. When you are finished, you can use the Copy Channel button to apply the same parameters to other channel.
SIP		
Call Transfer		
Enable	checkbox	Check this box to enable SIP Call Transfer for this channel. Both Blind and Assisted transfer are supported. For an Assisted call transfer, you would dial the transfer sequence, then at the dial tone, the "transfer to" number is dialed. After the "transfer to" endpoint answers, the dial transfer sequence is dialed again to complete the transfer. For a Blind call transfer, you dial the transfer sequence with an extra asterisk at the end. At the dial tone, dial the number for the "transfer to" endpoint. When the "transfer to" endpoint answers, you will automatically be disconnected and the call will automatically be transferred.
Transfer Sequence	characters	Enter the characters to be used as the call transfer sequence. A maximum of 4 characters may be used.
Call Hold		
Enable	checkbox	Check this box to enable SIP Call Hold for this channel.
Hold Sequence	characters	Enter the characters to be used as the call hold sequence. A maximum of 4 characters may be used.
GSM		
Call Transfer		
Enable	checkbox	Check this box to enable GSM Call Transfer for this channel.
Transfer Sequence	characters	Enter the characters to be used as the call transfer sequence. A maximum of 4 characters may be used.
Call Hold		
Enable	checkbox	Check this box to enable GSM Call Hold for this channel.
Hold Sequence	characters	Enter the characters to be used as the call hold sequence. A maximum of 4 characters may be used.
Call Forward		
Enable	checkbox	Check this box to enable Call Forwarding for this channel.
Forward Destination	number	Enter the phone number that a call meeting any of the conditions below will be forwarded to when flagged.
Conditions		
Unconditional	checkbox	Check this box to enable Unconditional call forwarding for this channel.
Not Reachable	checkbox	Check this box to enable Not Reachable monitoring for this channel. This could mean that the SIM number is disabled or the signal strength is too low.
Busy	checkbox	Check this box to enable Busy monitoring for this channel.
No Answer	checkbox	Check this box to enable No Answer monitoring for this channel. A duration time can be set with the No Answer condition; see below.
Duration	time	Duration is applicable to the 'No Answer' setting only. Select the time (in seconds) from the drop-down list available that you want the unit to wait until a No Answer state is flagged. Range is 5 -30 in (5 second increments).
OK	button	Clicking the OK button will retain any changes made and exit to the About screen.
Cancel	button	Clicking the Cancel button will discard all changes and exit to the About screen.
Copy Channel	button	Clicking Copy Channel will open a window allowing you to copy these setting to the other channel.
Default	button	Clicking the Default button will revert all sections to their factory default settings.
Help	button	This will open any help that may be available. Not all screens have an associated help file.

Call Routing

CLIP Routing

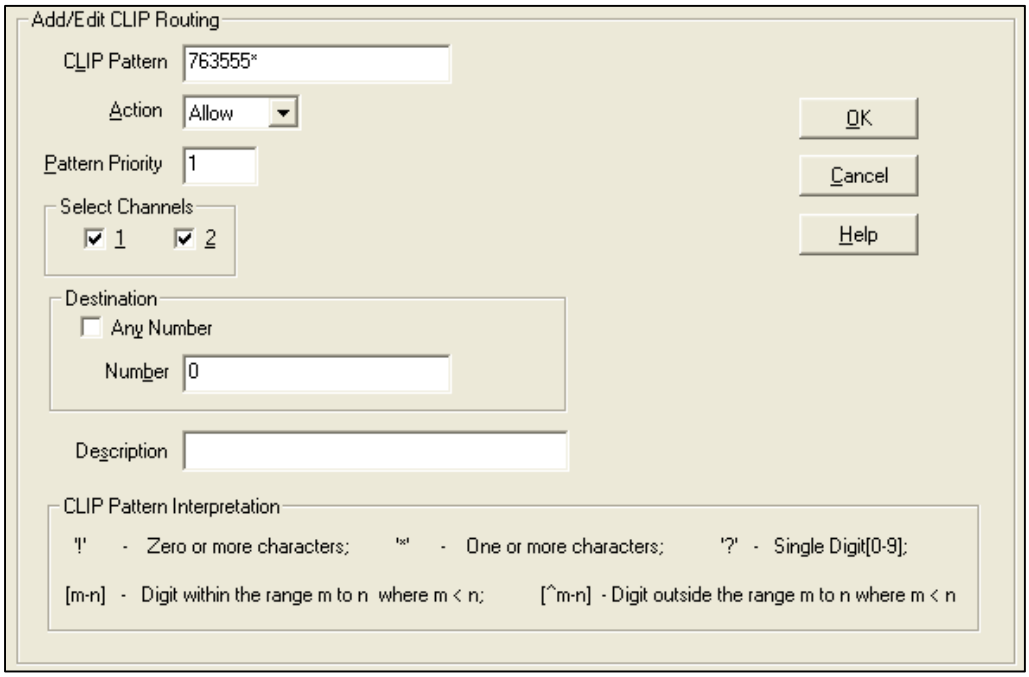
Calling Line Identification Presentation (CLIP) routing is a supplementary GSM service used to show the number of a caller.

Using CLIP routing, calls are accepted or rejected based on the calling number or a portion thereof. The incoming call is analyzed for pattern matches to CLIP routing entries. If there is no match, then the "Reject Calls Not Listed" checkbox is used to determine the processing of the call. If the box is checked, any non-matching calls are rejected. Left unchecked, calls then are routed according to any Auto-call settings. If there are no Auto Call settings present, then the incoming call is presented with a dial tone and the MVPGSM waits for digits to be dialed by the incoming user.



CLIP Routing parameters		
Field Name	**Values**	**Description**
Reject Calls Not Listed	checkbox	Select this if you want calls not matching a CLIP routing pattern to be rejected.
CLIP Pattern	symbols and numbers	Displays the CLIP pattern for this specific entry based on the interpretation table: **!** *Zero or more characters* ***** *One or more characters* **?** *A single digit (0 through 9)* **[m-n]** *A digit within the range entered, with digit 'm' less than digit 'n'.* **[^m-n]** *A digit outside of the range entered, with digit 'm' less than digit 'n'.*
Channels	number	Displays the channels using this routing pattern.
Pattern Priority	number	Show the priority number of this routing entry.
Destination Number	number	Displays the destination number matched to an outbound phone book entry.
Action	text	Display the action taken when the CLIP routing is triggered.
Description	text	Displays a description of this routing pattern.
Number of Entries	number	Display s the total number of CLIP routing entries for this channel.
Add	(button)	Opens the Add/Edit CLIP Routing screen.
Edit	(button)	Opens the Add/Edit CLIP Routing screen.
Delete	(button)	Clicking this button will permanently delete the routing entry.

Add/Edit CLIP Routing



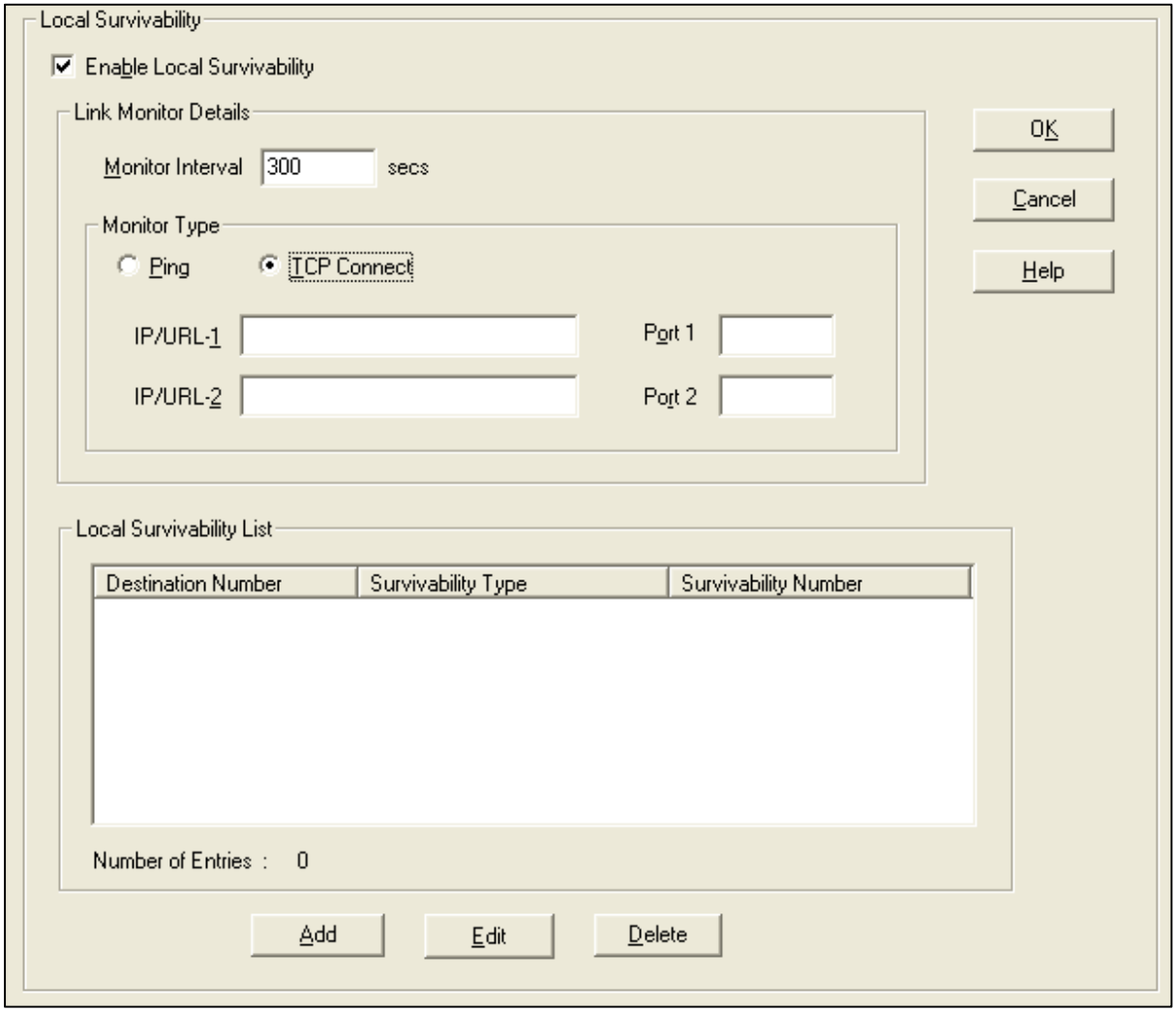
Add/Edit CLIP Routing parameters		
Field Name	**Values**	**Description**
CLIP Pattern	varies	Enter a valid CLIP Pattern (variation of a phone number or a specific phone number) here. Wildcards may be added as dictated by the CLIP Pattern Interpretation legend.
Action	Allow or Reject	Select the action to be applied to calls that match the CLIP pattern. Your options are to allow, which would then direct the call to a matching Outbound Phone Book entry, or reject, which would deny the incoming call from being passed on.
Pattern Priority	1 – 255	Select the priority of this pattern (1 is the highest priority).
Select Channels	checkboxes	Select all channels that will use this CLIP routing setting.
Any Number	checkbox	When 'Any Number' is checked, the incoming GSM caller will hear dial tone from the VoIP and must dial a number that matches an outbound phone book entry for the call to proceed over IP (this will gray out the Number box).
Number	number	Enter the destination number corresponding to an outbound phone book entry.
Description	varies	Enter a description that will help you remember what this pattern does.
CLIP Pattern Interpretation	!, *, ?, [m n], [^m n]	Essentially, these characters are used as wildcards with varying degrees of range or specifics. **!** – This allows for zero or more characters to be required. [This could allow for numbers that reported with or without a leading 1 for example.] ***** – One or more characters are allowed with this setting. [As in the example in the Phone Book section, this wildcard would expect at least one more character before triggering.] **?** – This is used in conjunction with a single number (0-9). [A single question mark or multiple question marks can be used to replace individual digits.] **[m-n]** – To specify a range of numbers, brackets are used and the numbers used to replace 'm' and 'n' must be in numeric order (*i.e.* 'm' must be less than 'n'). **[^m-n]** – To specify digits outside of a given range, brackets and a caret are used and the numbers used to replace 'm' and 'n' must be in numeric order (*i.e.* 'm' must be less than 'n').

Local Survivability

Local Survivability works in two ways:

IP Survivability will see if the IP link (WAN/LAN) is up, if not the call will be routed using the Outbound Phone Book with a new survivability number configured (where the IP address could be on the local network or a sub network address).

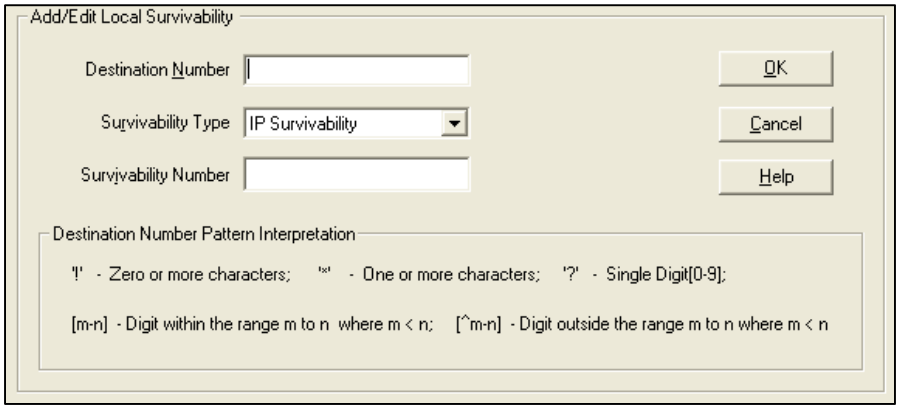
Internal Survivability will see if the IP link (WAN/LAN) is up and if not, it will route the call according to the settings in the Inbound Phone Book with a new survivability number configured (provides GSM failover).



Local Survivability continued:

Local Survivability parameters		
Field Name	**Values**	**Description**
Enable Local Survivability	(check box)	Check this box if you want your MVPGSM to use local survivability (route calls when certain connections are not available).
Link Monitor Details		
Monitor Interval	number of seconds	Enter the time (in seconds) for survivability monitoring to occur.
Monitor Type	Ping or TCP Connect	Select the type of monitoring to be done (by Ping or by TCP connections to a website). Ping is grayed out unless DNS is enabled in the Ethernet/IP configuration page.
IP / URL-1	URL	Enter an Internet / IP address that will respond to a TCP connect / PING request.
Port 1	number	This is the port number used for the above TCP connect attempt.
IP / URL-2	URL	Enter a second Internet address that will respond to a TCP connect / PING request.
Port 2	number	This is the port number used for the above TCP connect attempt.
Local Survivability List		
Destination Number		The number that corresponds to the Outbound phone book entries for routing.
Survivability Type	IP -or- Internal	IP Survivability or Internal Survivability. *IP Survivability* will see if the IP link (WAN/LAN) is up, if not the call will be routed using the Outbound Phone Book. *Internal Survivability* will see if the IP link (WAN/LAN) is up and if not, it will route the call according to the settings in the Inbound Phone Book.
Survivability Number		*IP Survivability:* an IP call will be placed based on this number using the Outbound phone book. *Internal Survivability:* a call will be placed (out another GSM channel) based on this number using the Inbound phone book.
Number of Entries		The total number of entries for survivability mode.
Add	(button)	Opens the Add/Edit Local Survivability screen.
Edit	(button)	Opens the Add/Edit Local Survivability screen.
Delete	(button)	This will permanently delete the selected entry.

Add/Edit Local Survivability List


Add/Edit Local Survivability
Destination Number
Survivability Type
IP Survivability
Survivability Number
OK
Cancel
Help
Destination Number Pattern Interpretation
'!' - Zero or more characters; '*' - One or more characters; '?' - Single Digit[0-9];
[m-n] - Digit within the range m to n where m < n; [^m-n] - Digit outside the range m to n where m < n

Field	Description
Destination Number	The number that corresponds to the Outbound phone book entries for routing.
Survivability Type	IP Survivability or Internal Survivability. *IP Survivability* will see if the IP link (WAN/LAN) is up, if not the call will be routed using the Outbound Phone Book. *Internal Survivability* will see if the IP link (WAN/LAN) is up and if not, it will route the call according to the settings in the Inbound Phone Book.
Survivability Number	*IP Survivability:* an IP call will be placed based on this number using the Outbound phone book. *Internal Survivability:* a call will be placed (out another GSM channel) based on this number using the Inbound phone book.

Save Settings

Save & Reboot

Saving the MultiVOIP GSM Configuration. When values have been set for all of the various operating parameters, click on **Save Setup** in the sidebar, then *Save & Reboot*.

Creating a User Default Configuration. When a "Setup" (complete grouping of parameters) is being saved, you will be prompted about designating that setup as a "User Default" setup. A User Default setup may be useful as a baseline of site-specific values to which you can easily revert. Establishing a User Default Setup is optional.

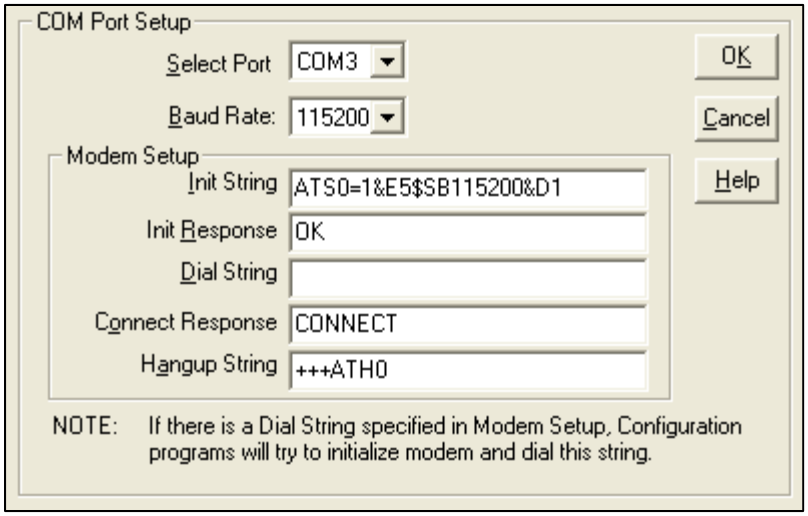
Connection

Settings

This is also accessible from the Start menu in the MultiVOIP GSM software folder.

Set Baud Rate. The **Connection** option in the sidebar menu has a "Settings" item that includes the baud-rate setting for the COM port of the computer running the MultiVOIP GSM software.

First, it is important to note that the default COM port established by the MultiVOIP GSM program is COM1. *Do not accept the default value until you have checked the COM port allocation on your PC*. To do this, check for COM port assignments in the system resource manager of your Windows operating system. If COM1 is not available, you must change the COM port setting to a COM port that you have confirmed as being available on your PC.

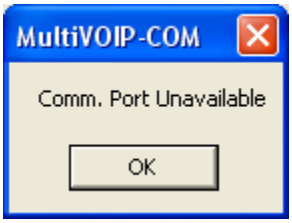


Troubleshooting Software Issues

In the lower left corner of the screen, the connection status of the MultiVOIP GSM will be displayed. The messages in the lower left corner will change as detection occurs. The message “MultiVOIP GSM Found” confirms that the MultiVOIP GSM is in contact with the MultiVOIP GSM configuration program. If the message displayed is “MultiVOIP GSM Not Found!” please try the resolutions below.

Fixing a COM Port Problem

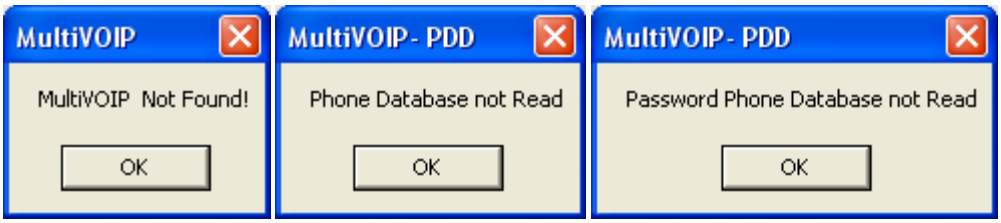
If the MultiVOIP GSM main screen appears but is grayed out and seems inaccessible, the COM port that was specified for its communication with the PC is unavailable and must be changed. An error message will appear.



To change the COM port setting, use the **COM Port Setup** dialog box, by going to the **Connection** pull-down menu and choosing “Settings” or use the left side control panel. In the “Select Port” field, select a COM port that is available on the PC (if no COM ports are currently available, re-allocate COM port resources in the computer’s MS Windows operating system to make one available).

Fixing a Cabling Problem

If the MultiVOIP GSM cannot be located by the computer, three error messages will appear (saying “Multi-VOIP Not Found”, “Phone Database Not Read” and “Password Phone Database Not Read).



In this case, the MultiVOIP GSM is disconnected from the PC running the MultiVOIP software – the cause of which may be a bad or unplugged cable, the VOIP is turned off or the COM port is wrong or bad. For instructions on MultiVOIP GSM cable connections, see the Cabling section of Chapter 3.

Chapter 5 – Phone Book Configuration

Introduction

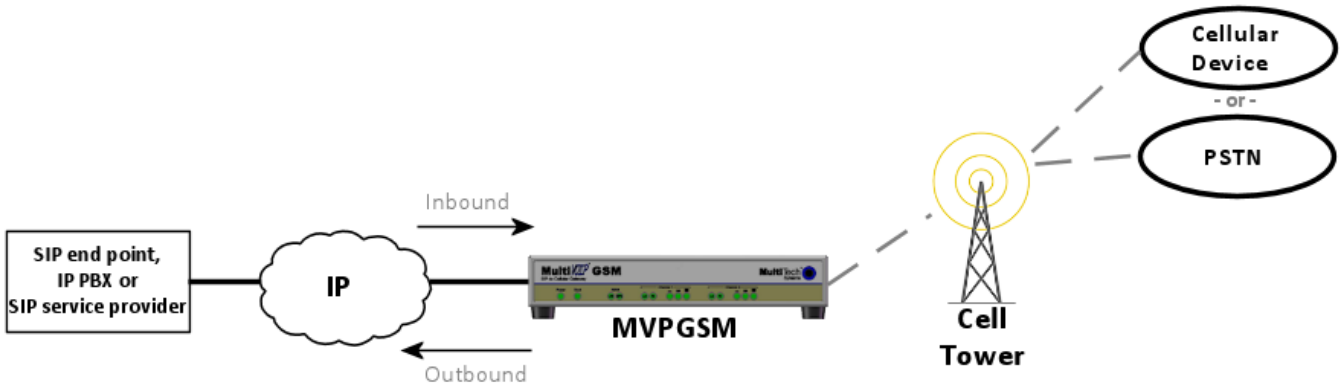
The wireless connectivity of the MVPGSM provides the cost-savings of wireless-to-wireless connections and inexpensive long distance to an existing location. By flagging calls that would connect to a remote site with another MultiVOIP unit or calls that are to be long distance, the MVP GSM can save money by routing those calls through the wireless connection instead of the standard PSTN. By configuring the phone books and setting up these routes, you can ensure that the calls placed are using the least cost possible to connect. The phone books also provide some security by limiting what numbers may be routed through them.

The "Inbound" phonebook is created with the dialing rules required for a call to be received on that VOIP. The "Outbound" phonebook describes dialing rules used to send calls from this VOIP, over IP, to a SIP end point. Inbound and Outbound refer to the connections that the MVPGSM has with devices that are accessible through IP, not those that connect through the wireless network.

Briefly stated, *the Outbound phone book lists the phone numbers (SIP end points) it will call; its Inbound phone book describes the dialing sequences that can be used to call into the MVPGSM and how those calls will be directed out to the cellular network.* The phone numbers are not literally "listed" individually, but are, instead, described by rule, such as an area code or beginning digit.

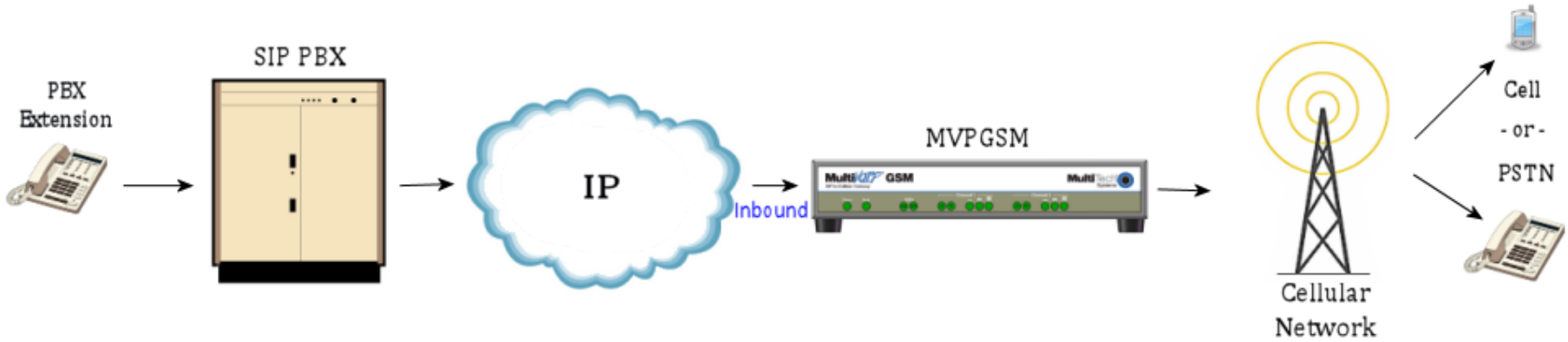
Note: Due to the differences in the technologies, we cannot guarantee that DTMF tones will be correctly translated from the cellular network.

Here is a basic setup example of how the MVPGSM can be put to use:

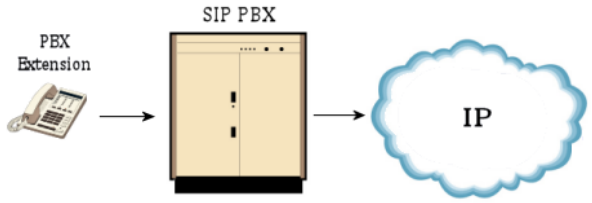


Sample Configuration: IP to Cellular Call Flow

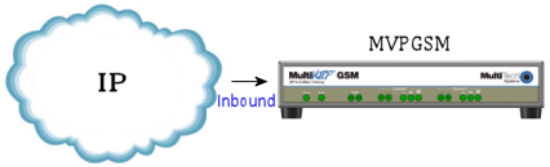
This sample section will show the process of making a call from a PBX extension and routing it to its end point using an MVPGSM. The graphic below represents the 'big 'picture' of the process; below that will be brief statements showing the steps of the process with a short statement describing the flow. Double arrows show the direction of flow within the statement.



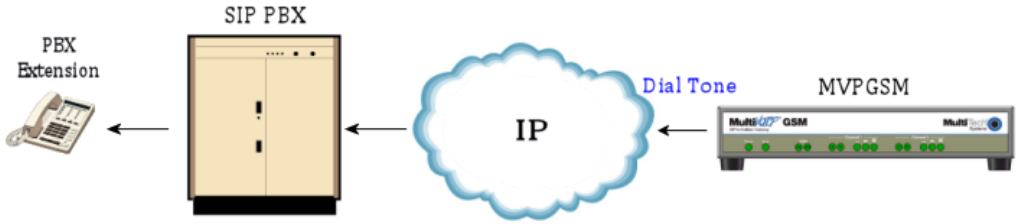
Call initiated from PBX Extension number 301 >> IP sends SIP invite packet with PBX extension 301:



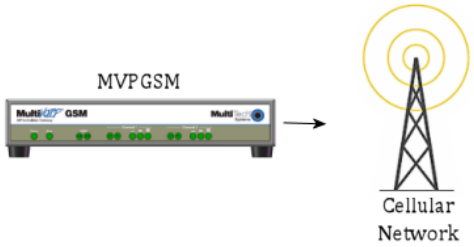
Inbound Phone Book removes prefix 301 >> call routed to available channel (channel will depend on Load Balancing settings):



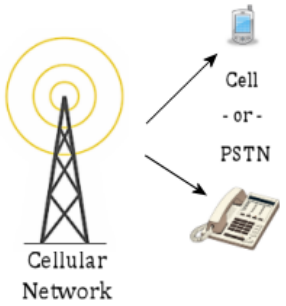
Dial tone in sent to PBX from << MVPGSM:



A user dials digits which are sent to the >> cellular network for call routing:



The call is placed from the cellular network to the intended >> end point (a cell phone, PSTN phone or PBX):

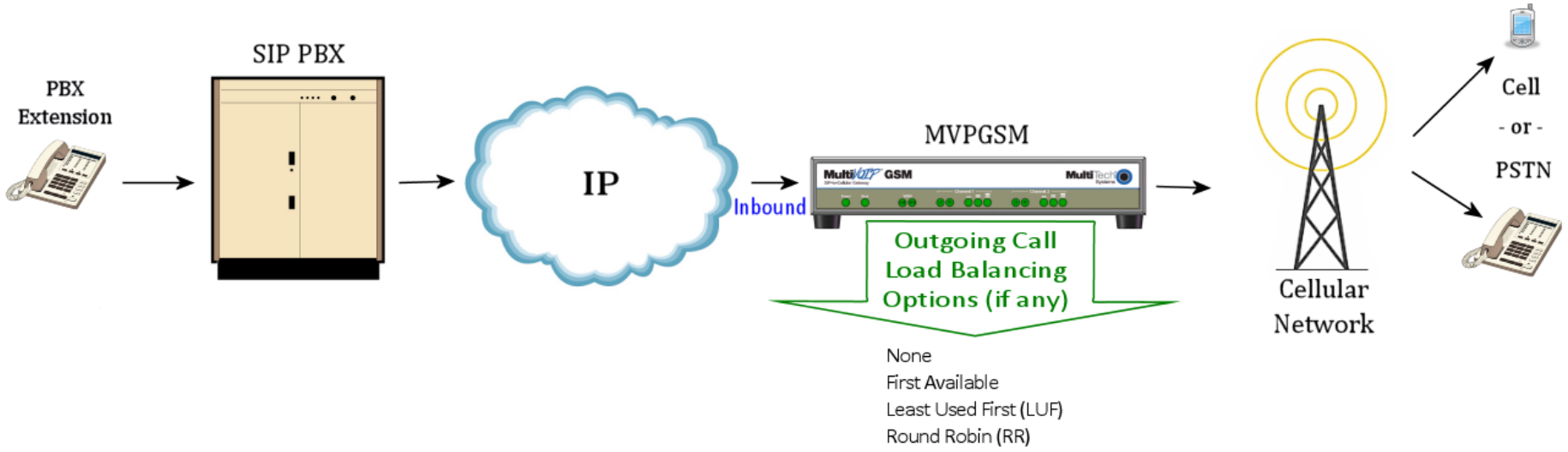


Call Forwarding

Call Forwarding is a feature of the Inbound Phone Book for IP calls and can be set to 'Unconditional' or 'Busy' as needed.
Unconditional forwarding will occur without any attempt to check the line status – an incoming IP call will be forwarded to another SIP end point over the IP network (the cellular network is not involved). See the Cellular to IP Call Flow section for call forwarding of cellular calls.
Busy forwarding will only occur if the line is detected as busy.

Making a Call: Load Balancing

Calls made through an MVPGSM unit can be routed according to the Load Balancing setting. There are three types of load balancing in the MVPGSM: First Available, LUF (Least Used First), and RR (Round Robin).



Load Balancing Options

Should you be using SIM cards that have limited service times, there are three load balancing styles that can be employed to maximize your use and reduce extra charges. If you have unlimited service, select 'Unlimited' in the load balancing screen. Load balancing is tied directly to the Inbound Phone Book. After creating a Load Balancing Group, you then need to go to the Inbound Phone Book and create the rules for the SIP end points that will take advantage of the load balancing group.

First Available:

First available balancing will attempt to use the first channel and if that is not available, it will attempt to use the next channel in numerical order.

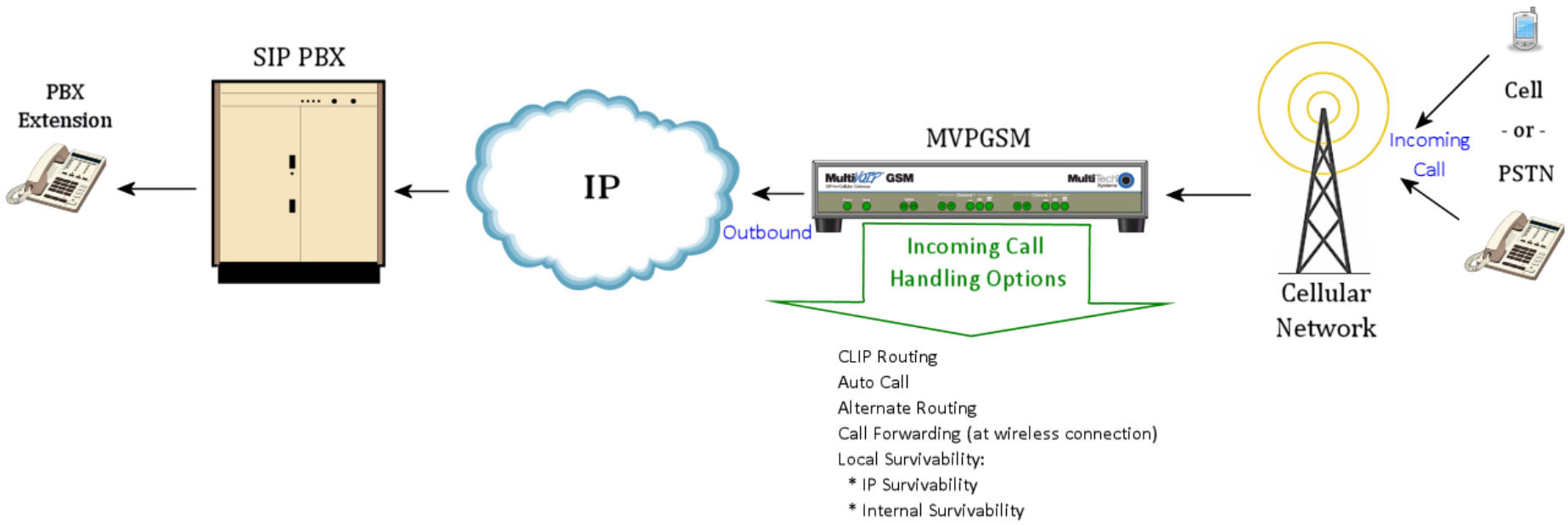
Least Used First:

Least Used First balancing will compare the calling plans on all channels and determine which has the highest number of minutes available per day and use that channel (SIM).

Round Robin:

Round Robin balancing will use a self-monitored list. The first channel used will be moved to the bottom of the list after it has been used once, and the next channel moves to the top of the list. This process rotates through all available channels set to Round Robin load balancing.

Sample Configuration: Cellular to IP Call Flow



Handling Options

There are several options available when handling cellular to IP calls; all of which can be stacked on top of each other (used together). Calls can also be routed based on the incoming dialed digits, but the reliability of these digits is not guaranteed due to the differences in technologies. The various options are described below.

CLIP Routing:

Calls are accepted or rejected based on the calling number. In the United States, this includes the leading "1" (so 555-123-4444 is the same as 1-555-123-444 for CLIP routing purposes). Accepted calls are routed over IP based on the CLIP routing reference to the Outbound Phone Book (destination pattern).

Auto Call:

Located in the Configuration | Voice/Fax section, Auto Call can be enabled with a number selected from the Outbound Phone Book. All calls received on this channel are then routed to the selected outbound destination.

Alternate Routing:

Found by clicking the 'Advanced' button for any Outbound Phone Book entry, Alternate Routing allows for an alternate IP address to be used for routing calls if the IP address in the Outbound Phone Book entry cannot be reached within the time defined by the 'Round Trip Delay' setting.

Call Forwarding (at the wireless connection):

Call Forwarding allows calls to roll from channel to channel if one or more are busy. In Configuration | Supplementary Services, you need to enable Call Forwarding.

Local Survivability:

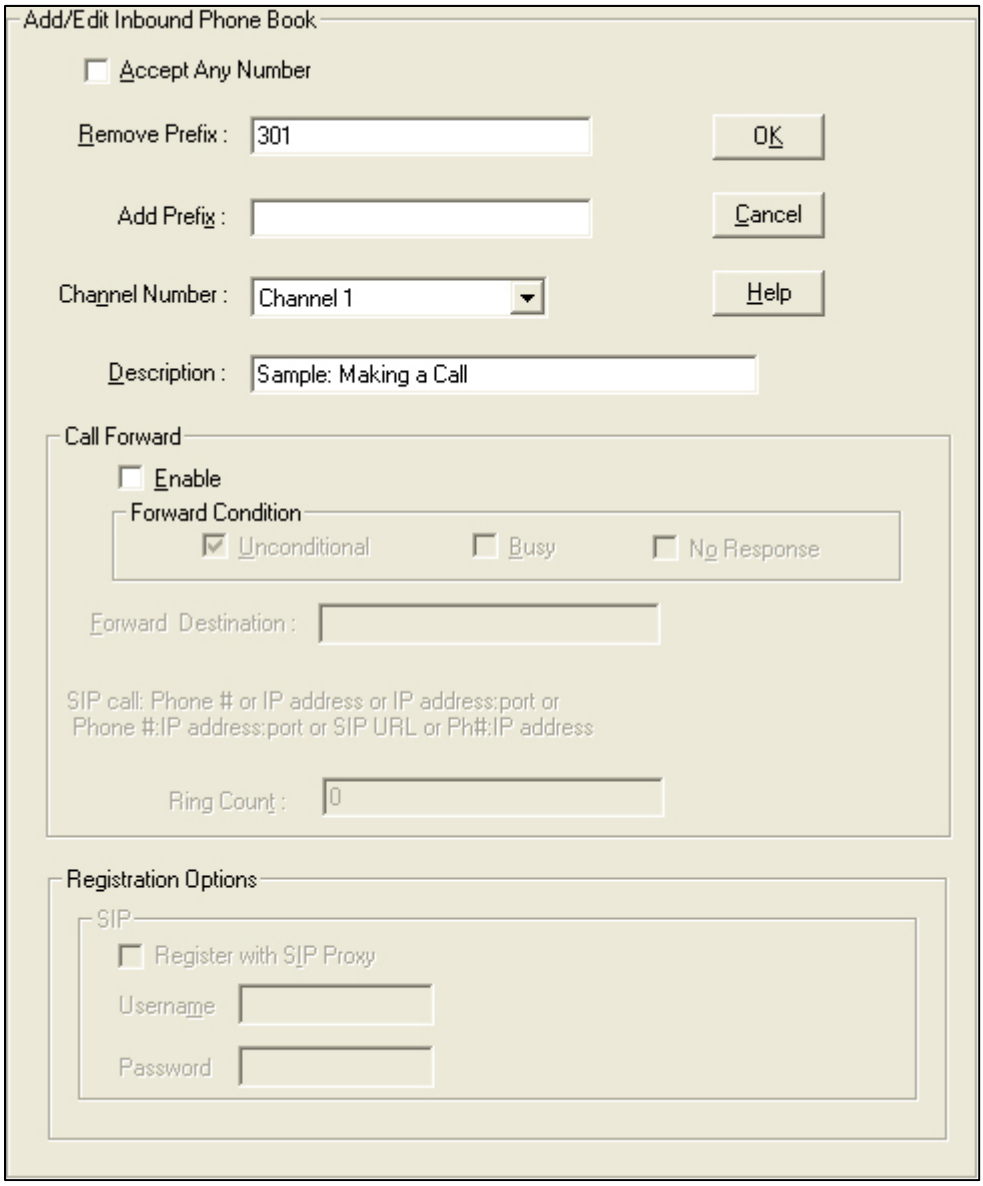
IP Survivability - If the IP link monitor destination cannot be reached, the call will be routed using the Outbound Phone Book.

Internal Survivability - If the IP link monitor destination cannot be reached, the call will be routed using the Inbound Phone Book.

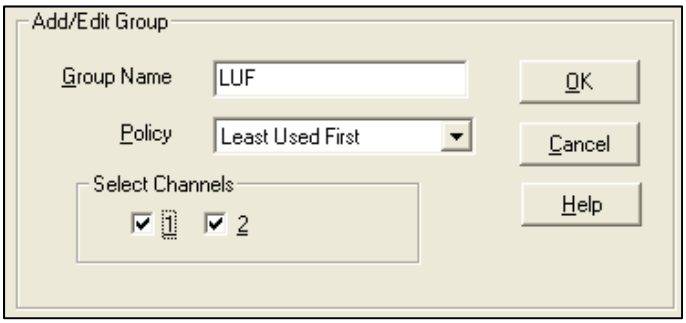
Sample Inbound Phone Book

The basic purpose of the Inbound Phone Book is to create rules for routing incoming IP calls. As in the samples previously shown, the phone books play an integral part in the call process.

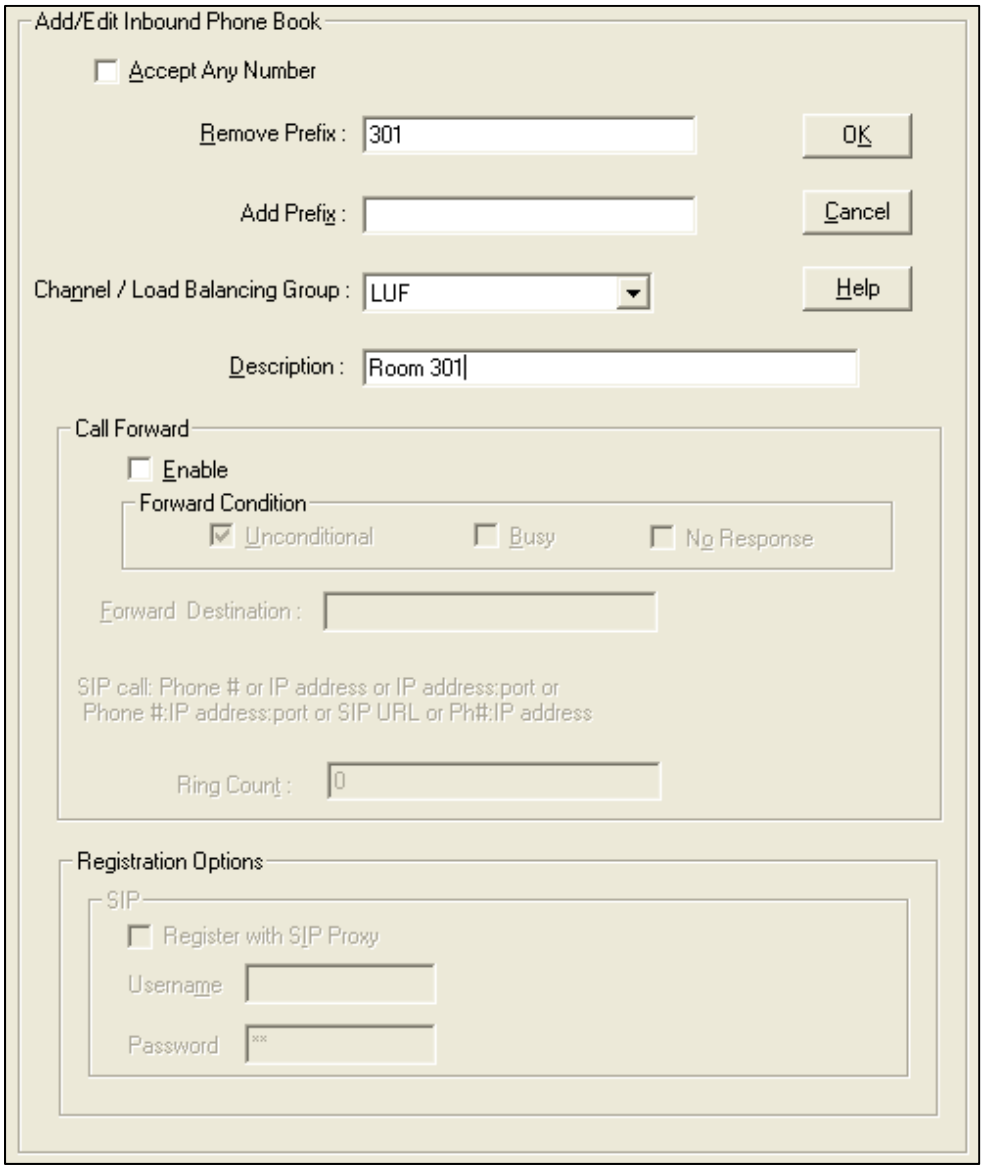
From the sample above for 'Making a Call,' we would have a basic Inbound Phone Book entry as shown below if that extension (extension number 301) was to use Channel 1:



If the SIMs have a calling plan other than unlimited, we would then apply load balancing as shown in the second sample 'Making a Call: Load Balancing.' For this we would create a Load Balancing Group as shown below. Here we will use LUF load balancing for the MVPGSM.



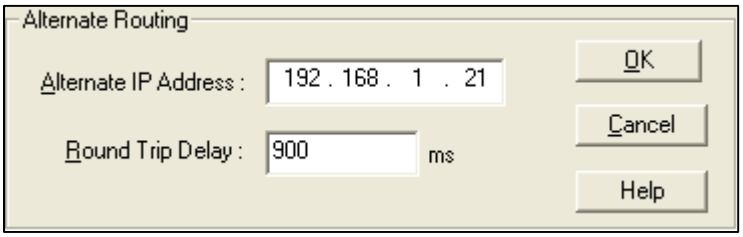
After creating this group, return to the Inbound Phone Book and in the entry we created for extension 301, use the 'Channel Number' drop-down box to select the newly created load balancing group 'LUF' and click OK. The group name can be any meaningful name that identifies the channels in the group.

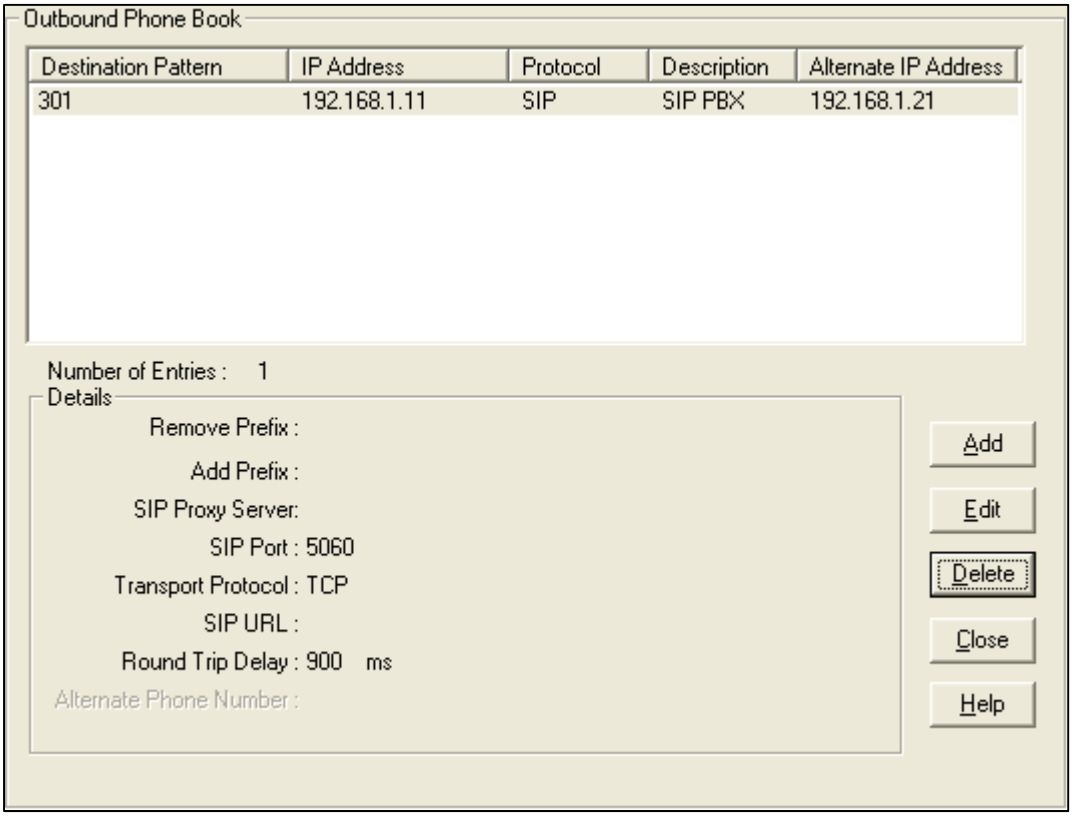


Sample Outbound Phone Book

The Outbound Phone Book is the director of where calls will be routed from the cellular network to the IP network – whether that is to a specific phone number, an available extension (Any Number) or to the IP address of a SIP device or remote VOIP unit for resending as a local call. As in the example previously shown, the phone books play an integral part in the processing of calls.

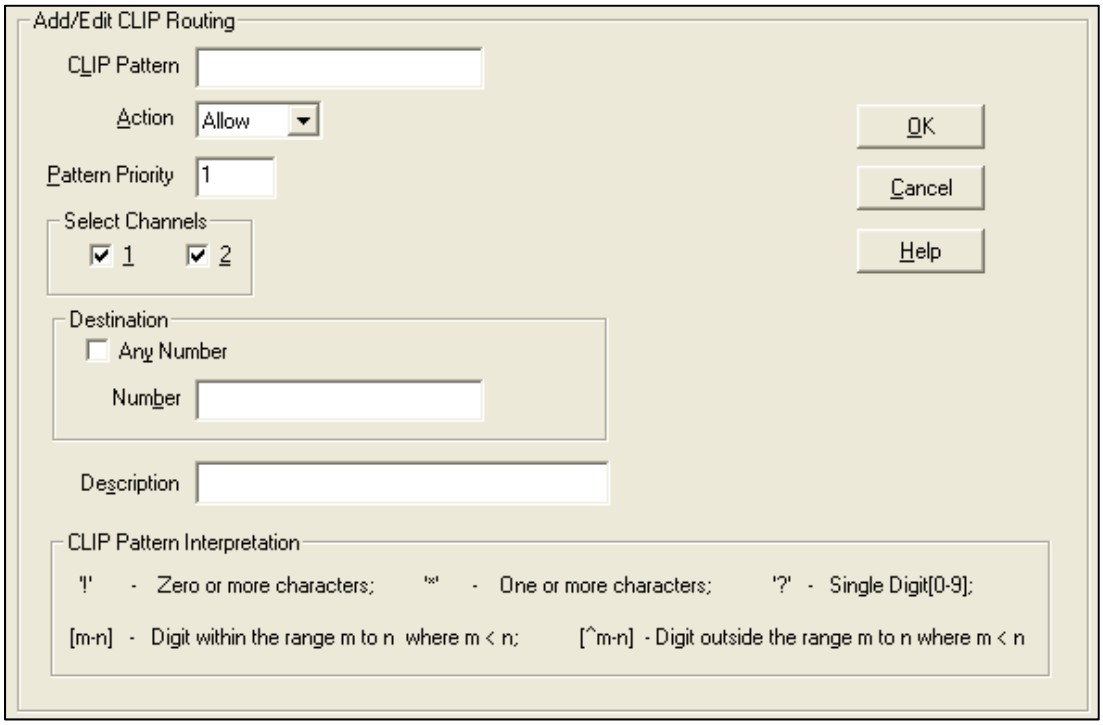
Below, we take advantage of Alternate Routing by clicking on the Advanced button for the Outbound Phone Book entry for our PBX Extension of 301. Then the IP of the alternate device is added to the phone book entry and the Round Trip Delay is set to 900 ms to allow extra time for the initial connection to attempt successful completion – once the round trip delay time has been passed without a successful connection, the call will be routed to the device using IP address 192.168.1.21 (as opposed to the primary device using 192.168.1.11).

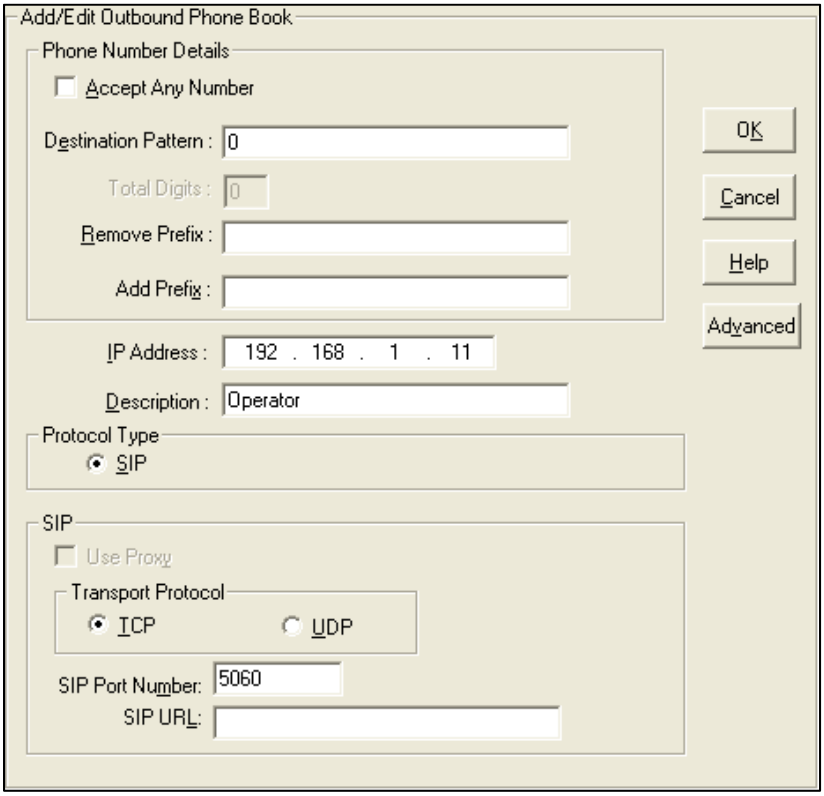




CLIP Routing Example

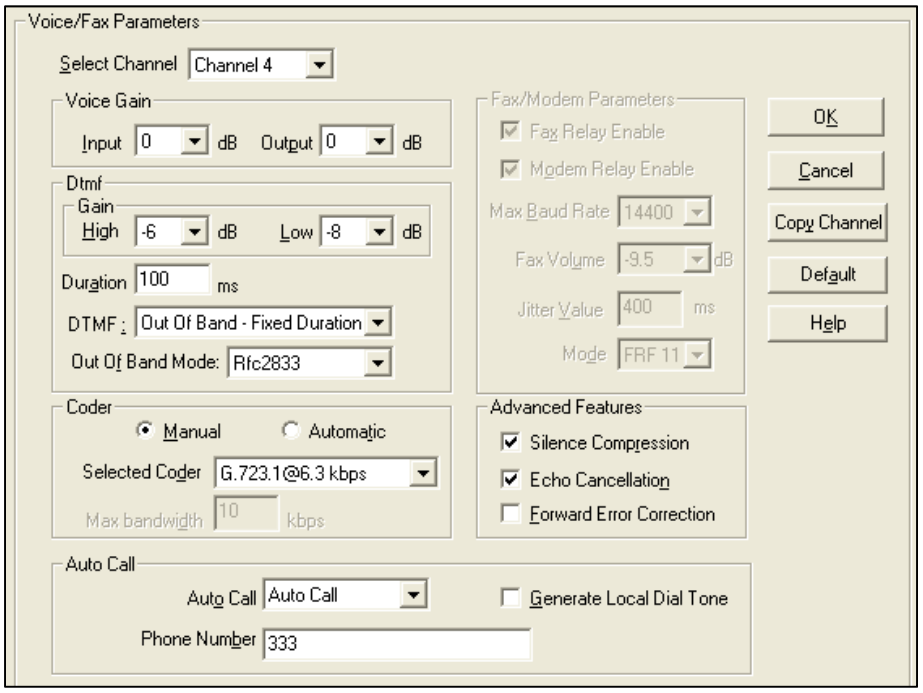
With CLIP routing, you can create rules or even specific numbers that will be routed using the Outbound Phone Book to specified recipients. As an example, we will set CLIP routing to take all calls from the 763 area code with a 555 prefix and have them directed to the operator on the local PBX. First, the CLIP routing screen is shown below. The *character is used in this case to allow “one or more characters” after the area code and local exchange number. After that, the corresponding Outbound Phone Book for the operator is shown.

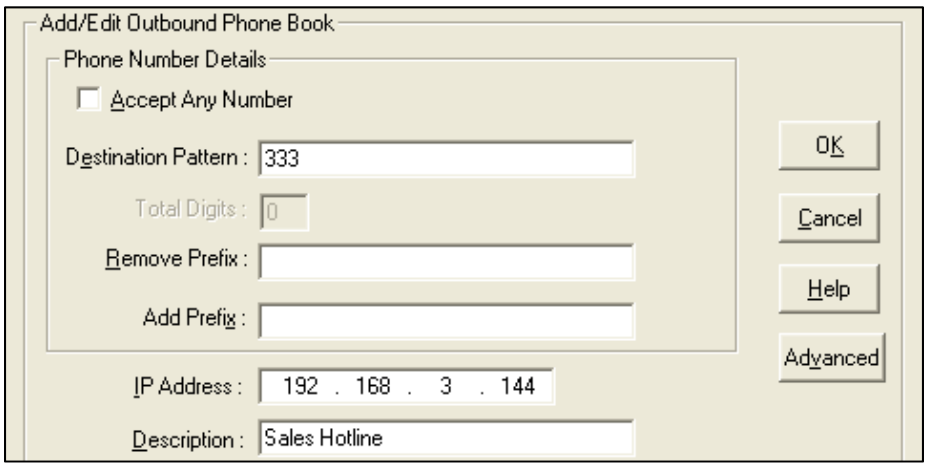


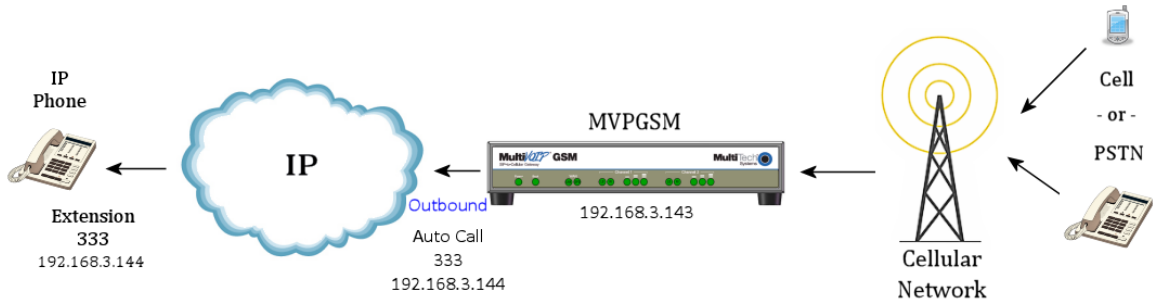


Auto Call Example

In cases where you want incoming GSM calls automatically routed, the Auto Call feature may be used to handle this. In this example, SIM on Channel 4 has a number of 763-555-4444 and we want all calls received by this number to go to the Sales department at extension 333. Open the Configuration | Voice/Fax screen and select the Channel that you want Auto Call enabled for (in this case, Channel 4) and in the Auto Call section, Set the Auto Call drop-down box to Auto Call and enter the Phone Number to be automatically dialed below (extension 333 in this case). Second, we need to create the corresponding Outbound Phone Book entry by adding the extension (333) in the Destination Pattern, the IP Address and a description for this entry (Sales Hotline).

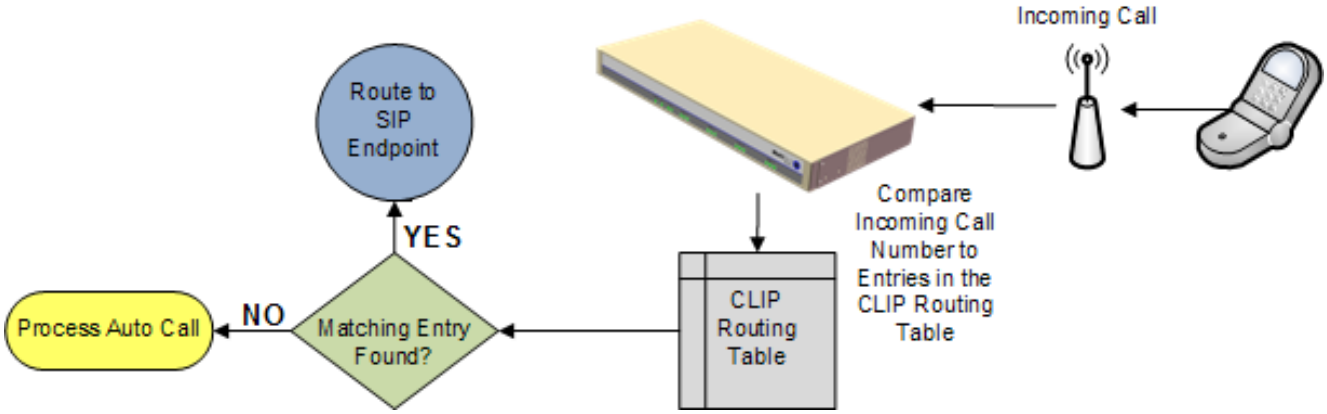






CLIP Routing used with Auto Call

Used together, CLIP Routing and Auto Call can handle incoming traffic efficiently. With a populated CLIP routing table, an incoming call will first be compared to all of the entries in the CLIP Routing table looking for a match (full number or partial as the rules are set) and if no match is found, Auto Call will route the call according to the Auto Call settings. This way, incoming calls are routed as specified by CLIP Routing and those that do not are routed to a number that is purposed for such calls.



For example, if your MVPGSM had a CLIP routing table setup to route calls from the 763 area code to the main office (Ext. 320) and calls from the 651 area code to go to the accounting department (Ext. 322) and calls from the 612 area code are sent to maintenance (Ext. 390), all calls from other area codes would be subject to the Auto Call settings – in this case, we want those to go to a receptionist (Ext. 200). A CLIP routing pattern is added for each of the three area codes. Set all channels to Auto Call extension 200. Then the corresponding Outbound Phone Book entries are created (you will actually be prompted after each CLIP pattern entry to create a Phone Book entry). The entries are shown below.

CLIP Routing

☐ Reject Calls Not Listed

CLIP Pattern	Channels	Pattern Priority	Destination Number	Action	Description
763*	1,2	1	320	Allow	Main Office
651*	1,2	1	322	Allow	Accounting
612*	1,2	1	390	Allow	Maintenance

Auto Call

Auto Call: Auto Call ☐ Generate Local Dial Tone

Phone Number: 200

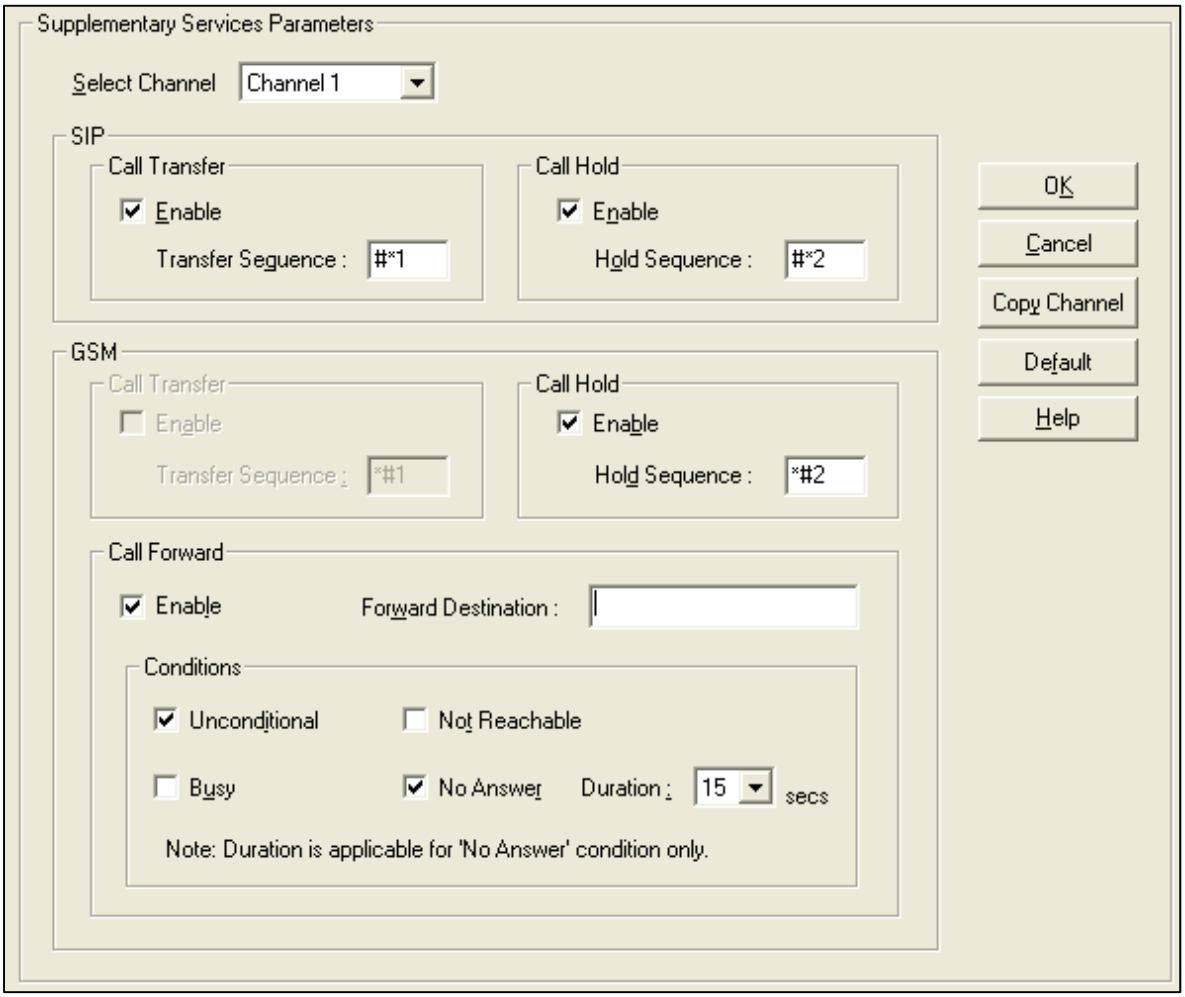
Outbound Phone Book

Destination Pattern	IP Address	Protocol	Description	Alternate IP Address
200	192.168.3.171	SIP	Receptionist	
320	192.168.3.150	SIP	Main Office	
322	192.168.3.151	SIP	Accounting	
390	192.168.3.160	SIP	Maintenance	

Call Forwarding Example

Another call handling option will be used in this example: Call Forwarding. This sets the channels to forward on a 'busy,' 'no reply' and 'not reachable' signal and to disable the call waiting feature. This is accomplished by checking the Call Forward box to 'Enable' as shown below. You then need to add a phone number in the Forward Destination field.

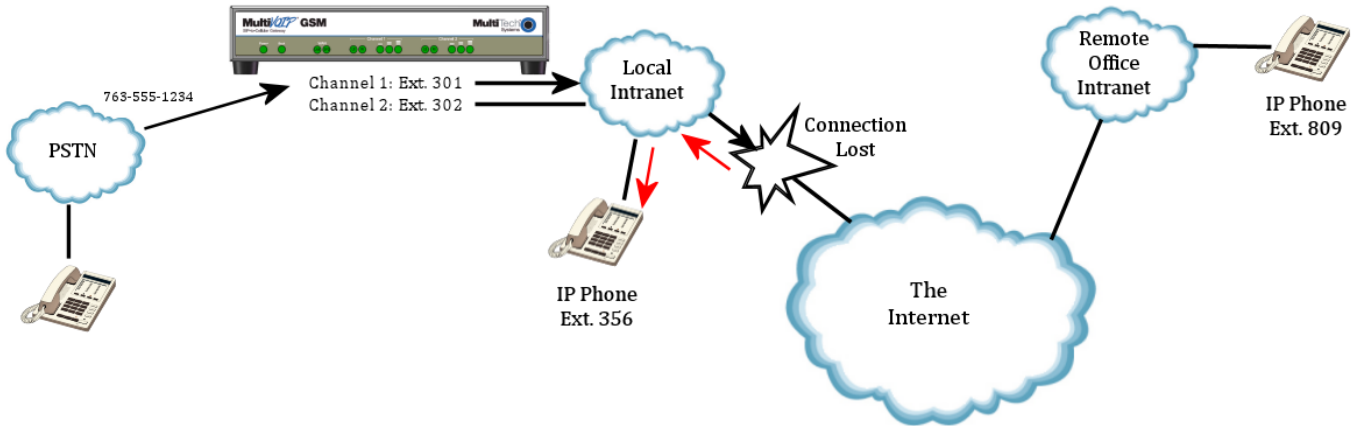
By setting each channel to forward to the next available until the last channel is reached, just one number can be given out to call into the MVPGSM from the cellular network.



Local Survivability

In cases where the connectivity to the network is not available (no response received from the IP Address/URL set in Local Survivability – Link Monitor Details), Local Survivability can route calls strictly through the available entries found in the Inbound and Outbound Phone Books. *IP Survivability* utilizes the Outbound Phone Book for call routing while *Internal Survivability* uses the Inbound Phone Book.

IP Survivability Example: You have an MVPGSM with Channel 1 using SIP extension 301 and Channel 2 using SIP extension 302. A GSM call is received on Channel 1 and is intended for a remote office (809) represented by the black arrows. Unfortunately, the survivability destination on the Internet is not responding, so Local Survivability mode is processed, represented by the red arrows. With the survivability link monitor reporting no connectivity, IP Survivability routes the call to the IP Phone at extension 356 as this is the route set for survivability. There needs to be an Outbound Phone Book entry associated with this.



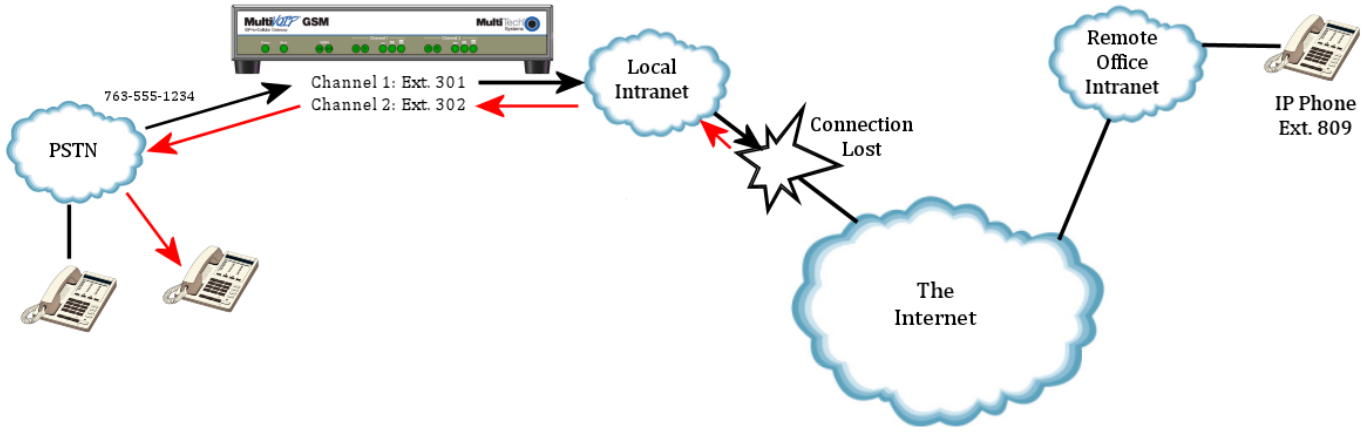
Local Survivability List

Destination Number	Survivability Type	Survivability Number
809	IP Survivability	356

Outbound Phone Book

Destination Pattern	IP Address	Protocol	Description	Alternat
200	192.168.3.171	SIP	Receptionist	
320	192.168.3.150	SIP	Main Office	
322	192.168.3.151	SIP	Accounting	
356	192.168.3.180	SIP	IP Survive	
390	192.168.3.160	SIP	Maintenance	

Internal Survivability Example: You have an MVPGSM with Channel 1 using SIP extension 301 and Channel 2 using SIP extension 302. A GSM call is received on Channel 1 and is intended for a remote office (809) represented by the black arrows. Unfortunately, the survivability destination on the Internet is not responding, so Local Survivability mode is processed, represented by the red arrows. With the survivability link monitor reporting no connectivity, Internal Survivability routes the call to channel 2 if the VOIP as this is the internal survivability route. There needs to be an inbound phone book entry for the survivability number. If the inbound entry (Inbound Phone Book entry '302' below) has a phone number in its Add Prefix field that number will automatically be called. If not (as shown below), the caller will hear dial tone from the GSM network and it can dial the desired destination.



Local Survivability List

Destination Number	Survivability Type	Survivability Number
809	Internal Survivability	302

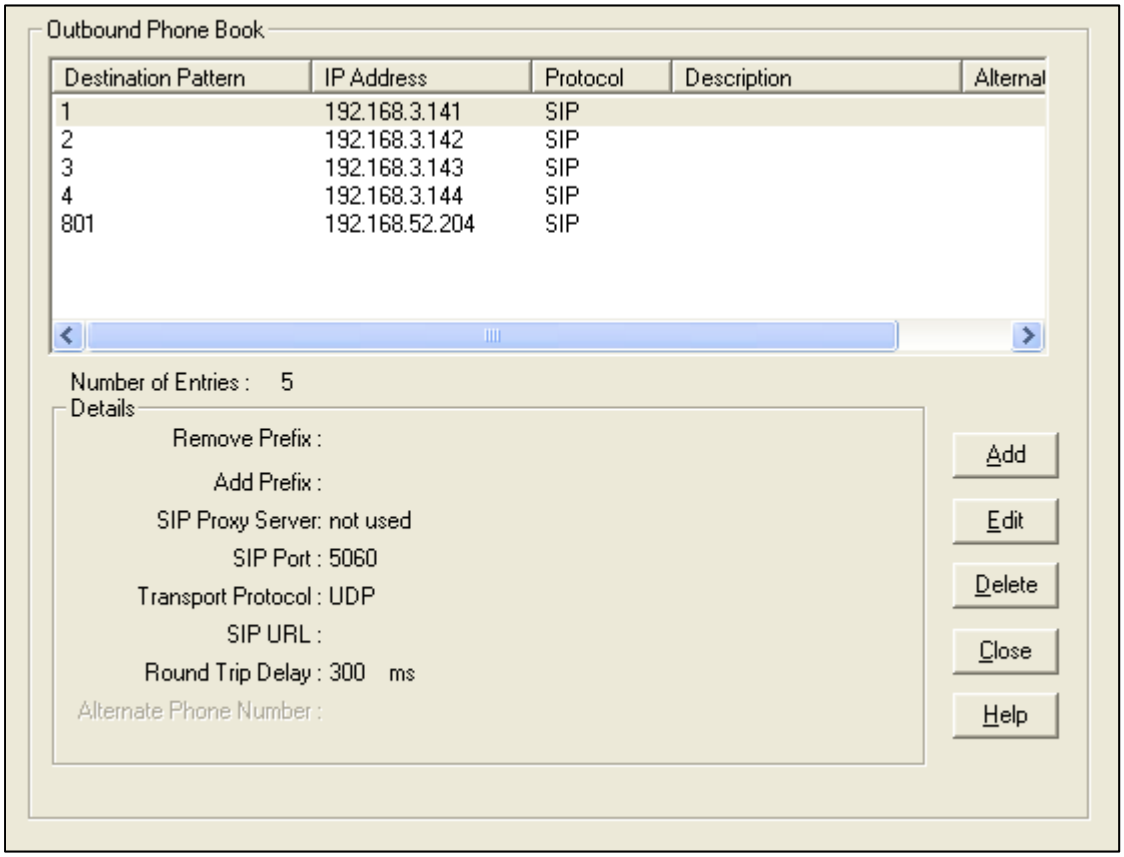
Inbound Phone Book

Remove Prefix	Add Prefix	Forward Address
301		Not Used
302		Not Used

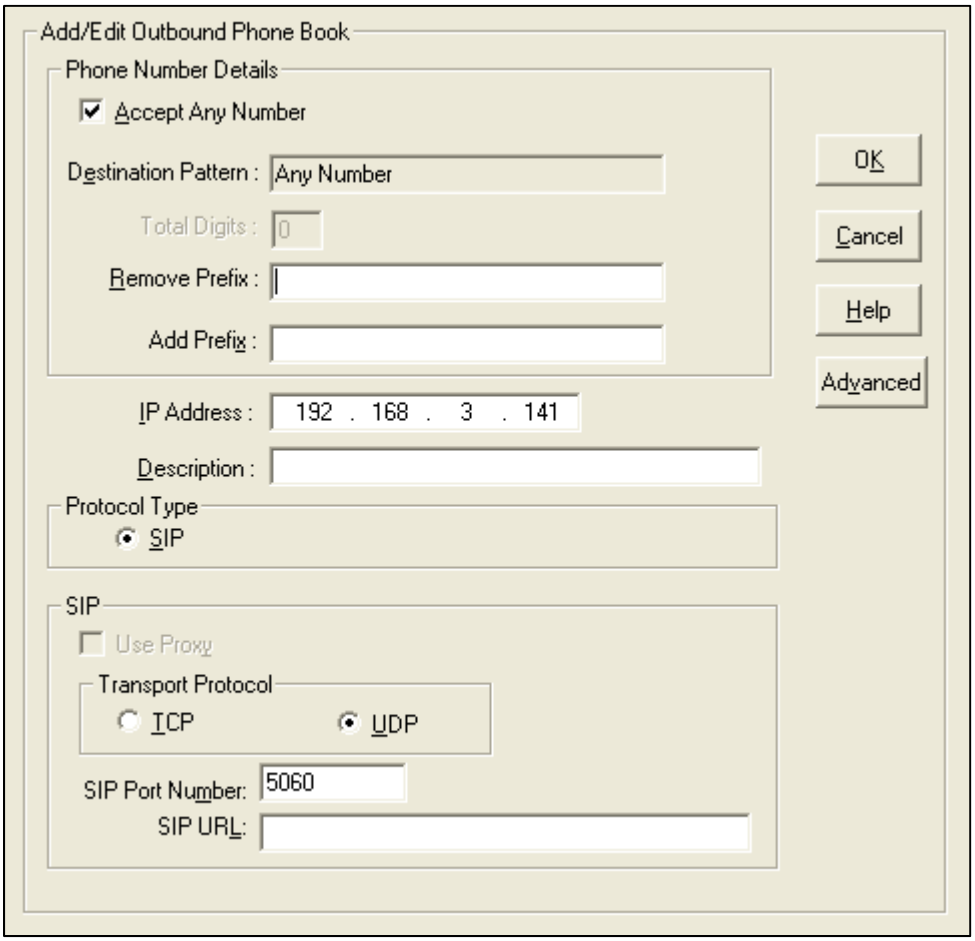
Phone Book Descriptions

Outbound Phone Book/List Entries

Some of the SIP-related fields in the “Details” section may be grayed out.



Add/Edit Outbound Phone Book



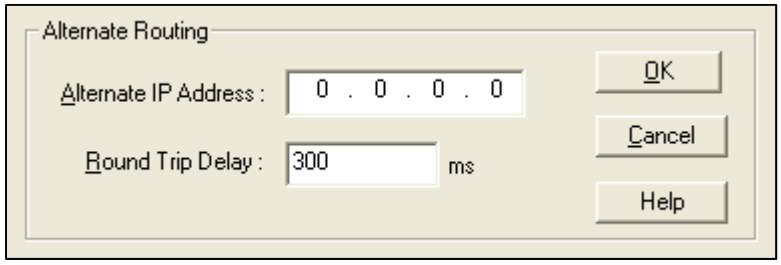
Enter Outbound Phone Book data for your MultiVOIP GSM unit. Note that the Advanced button gives access to the Alternate IP Routing feature, if needed. Alternate IP Routing can be implemented in a secondary screen (as described after the primary screen field definitions below).

The fields of the **Add/Edit Outbound Phone Book** screen are described in the table below.

Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	checkbox	When checked, "Any Number" appears as the value in the Destination Pattern field. The **Any Number** feature works differently depending on whether or not an external routing device is used (Proxy for SIP protocol). **When no external routing device is used**. If **Any Number** is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the **IP Address** in the Add/Edit Outbound Phone Book screen. "Any Number" can be used in addition to one or more Destination Patterns. **When external routing device is used**. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external routing device used (Proxy for SIP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be routed to a SIP end point on the IP network. This is the phone number or beginning portion of the phone number for calls that are to be routed to the IP address listed below.
Total Digits	as needed	Number of digits the phone user must dial to reach specified destination. *This field not used in North America*
Remove Prefix	dialed digits	Portion of dialed number to be removed before remaining digits are sent to their destination.
Add Prefix	dialed digits	Digits to be added before routing the call to the address below.
IP Address	*n.n.n.n*	The IP address to which the call above will be routed with the destination pattern given.
Description	alpha-numeric	This description will be sent as Caller ID information.
Protocol Type	SIP	Indicates protocol to be used in outbound transmission.

SIP Fields		
Use Proxy	checkbox	Select if proxy server is used.
Transport Protocol	TCP or UDP	VOIP administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC 3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The VOIP will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	*sip.userphone@hostserver*, where "userphone" is the telephone number and "hostserver" is the domain name or an address on the network	Looking similar to an email address, a **SIP URL** identifies a user's address. In SIP communications, each caller or callee is identified by a SIP URL: sip:user_name@host_name. The format of a sip URL is very similar to an email address, except that the "sip:" prefix is used.

Clicking on the **Advanced** button brings up the **Alternate Routing** secondary screen. This feature provides an alternate path for calls if the primary IP network cannot or does not respond within the timeframe of the Round Trip Delay. Often in cases of failure, call traffic is temporarily diverted into the PSTN. However, this feature could also be used to divert traffic to a redundant (backup) unit in case one SIP end point fails. The user must specify the IP address of the alternate route for each destination pattern entry in the Outbound Phonebook.

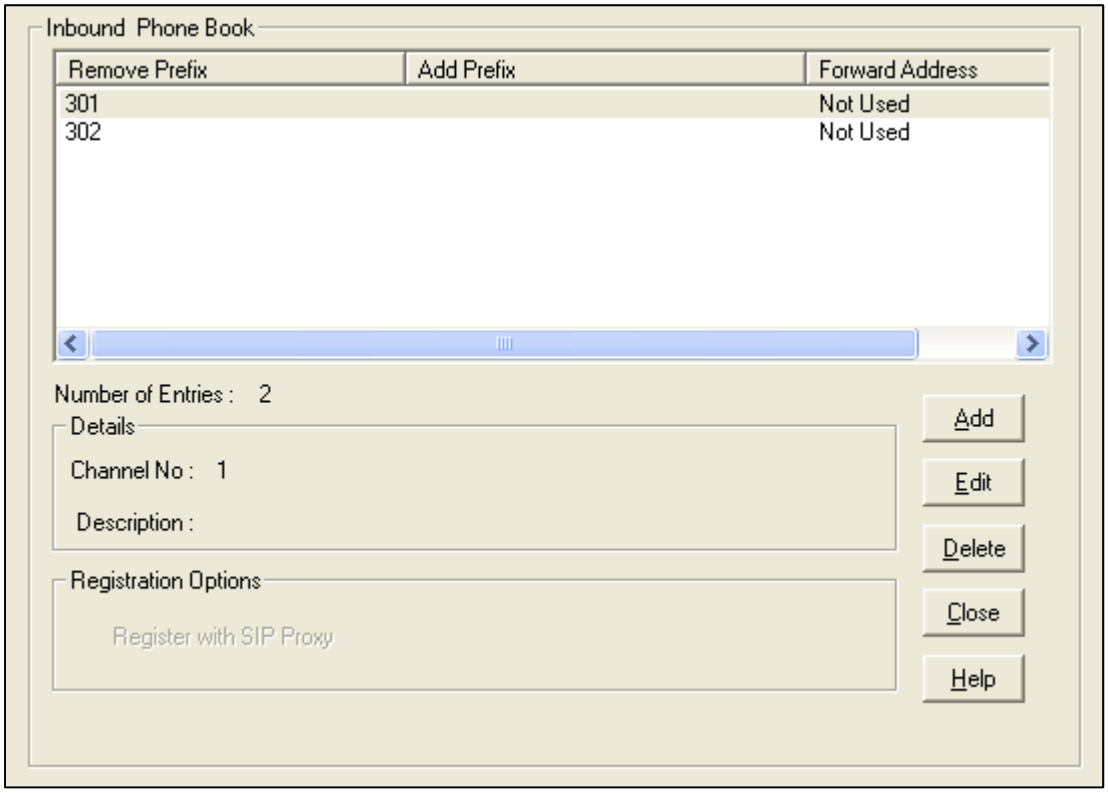


Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	*n.n.n.n*	Alternate destination for call traffic in case of excessive network delay.
Round Trip Delay	Default is 300 milliseconds	The Round Trip Delay is the criterion for judging when a data pathway is considered blocked. When the delay exceeds the threshold specified here, the data stream will be diverted to the alternate destination specified as the Alternate IP Address.

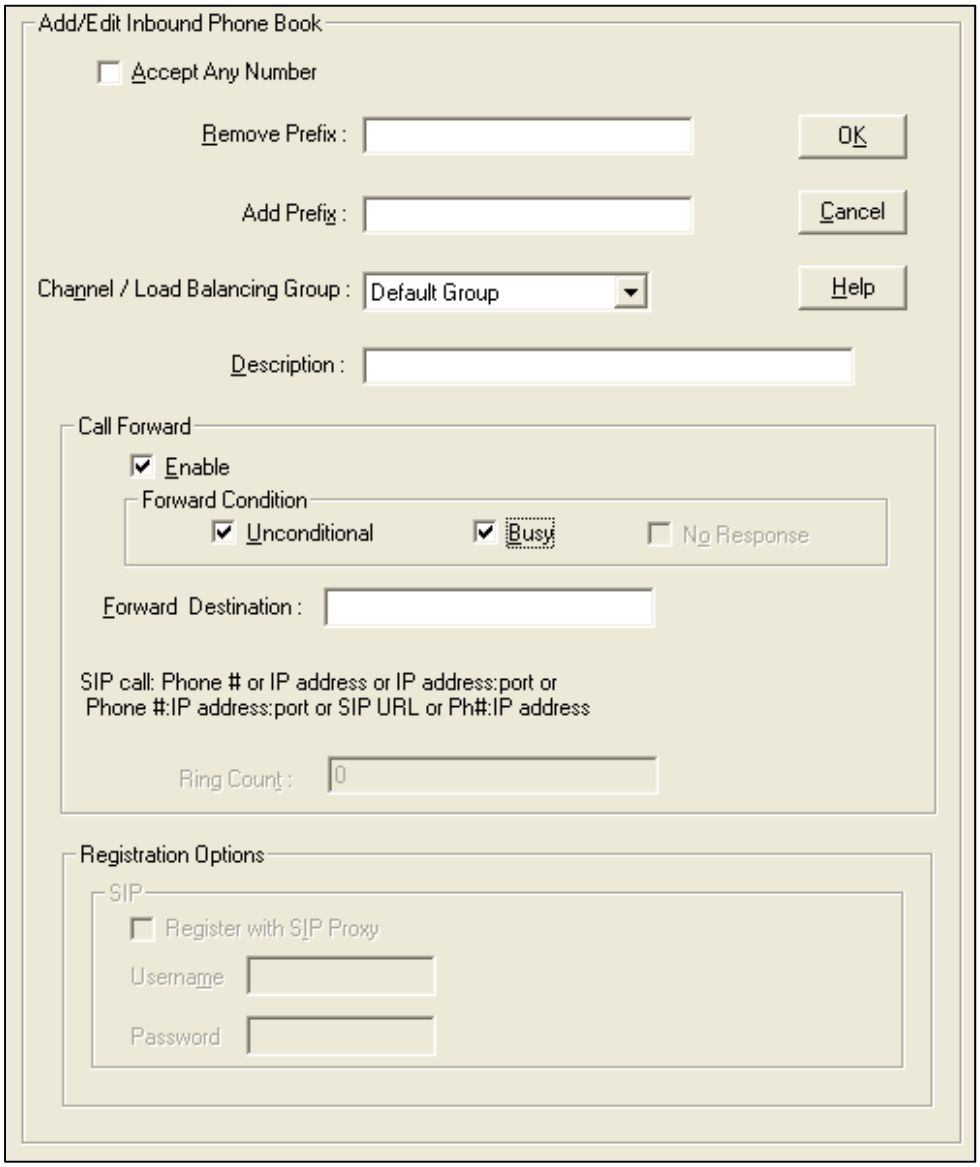
The Alternate Routing function facilitates PSTN Failover protection, that is, it allows you to re-route VOIP calls automatically if the VOIP system fails. The MultiVOIP GSM can be programmed to respond to excessive delays in the transmission of voice packets, which the MultiVOIP GSM interprets as a failure of the IP network. Upon detecting an excessive delay in transmission of voice packets (overly high “latency” in the network) the MultiVOIP GSM diverts the call to another of its channels, allowing the call to go back out to the wireless network to reach the remote end using a land line.

Inbound Phone Book/List Entries

The “Details” and “Registration Options” sections will display information based on the setup and protocols chosen.



Add/Edit Inbound Phone Book



Enter Inbound Phone Book data for your MultiVOIP GSM. The fields of the Add/Edit Inbound Phone Book screen are described in the table below.

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	checkbox	When checked, "Any Number" appears as the value in the Remove Prefix field. The **Any Number** feature of the Inbound Phone Book does not work when an external routing device is used (Proxy for SIP protocol). **When no external routing device is used**. If **Any Number** is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the VOIP on the channel listed in the **Channel Number** field. "Any Number" can be used in addition to one or more Prefixes.
Remove Prefix	dialed digits	Portion of digits to be removed from the incoming IP call before initiating a call on the wireless network using the remaining digits.
Add Prefix	dialed digits	Digits to be added to the incoming IP call after digits are removed (if any) but before initiating a call on the wireless network.
Channel Number	channel, Default Group, or other group	Wireless VOIP channel which the incoming IP call will be routed to. Default Group directs the call to any available channel and enables load balancing to keep one channel from being used more often than another. Any other group can be used, whether already in the unit or user defined.
Description	--	Optional description field.
Call Forward Parameters		
Enable	checkbox	Click the check-box to enable the call-forwarding feature.
Forward Condition	Unconditional -or- Busy	**Unconditional**. When selected, all calls received will be forwarded. **Busy**. When selected, calls will be forwarded when station is busy. No Response. This is a legacy selection that is not used by the MVPGSM.
Forward Destination	IP address, phone number, port number, etc	Phone number or IP address to which calls will be directed. For **SIP** calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number: IP address: port number, (e) SIP URL, or (f) phone #: IP address.
Ring Count		"No Response" type call forwarding is not supported. See 'Forward Condition' above.
Registration Option Parameters	In a **SIP** VOIP system, gateways can register with the SIP Proxy.	

Authorized User Name and Password for SIP

To enable the Registration Options on the Add/Edit Inbound Phone Book, you have to activate Use SIP Proxy Option on the Call Signaling, SIP Parameters Screen. Then add the IP address for the Primary Proxy in the SIP Proxy Parameters. This allows you to add a Username and Password to the Inbound Phone Book entry.

This feature is used when the MultiVOIP GSM registers with the proxies that support authorization and need the username, password and the endpoint name to be unique.

The VOIP sends Register request to Registrar for each entry with its configured Username and Password. When Authentication is enabled for the endpoint, then the registrar/proxy sends “401 Unauthorized/407 Proxy Authentication Required” response when it receives a REGISTER/INVITE request. Now, the endpoint has to send the authentication details in the Authorization header. In this header one of the fields is “username”.

Generally proxies accept requests even if both Endpoint Name and Username are same. But some proxies expect that the Endpoint Name and Username should be different.

To support these proxies, we have the username and password configuration for every inbound phone book entry which gets registered with a proxy.

If the username and password are not configured in the inbound phone book, then the registration will happen with the default username and password that are configured in the SIP Call Signaling Page.

Phone Book – Save and Reboot

When the Outbound and Inbound Phonebook entries are completed, click on **Save Setup** in the sidebar menu to save your configuration. You can change your configuration at any time as needed for your system.

Remember that the initial MultiVOIP GSM setup must be done locally using the MultiVOIP GSM program. After the initial configuration is complete, all of the MultiVOIP GSM units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVOIP GSM web interface software program or the MultiVOIP GSM program.

Chapter 6 – Using the Software

Introduction

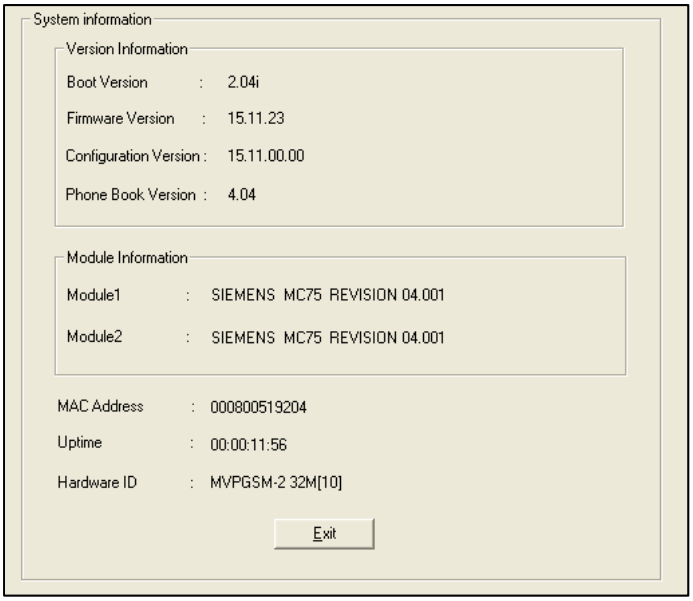
This chapter will primarily cover the day to day operation and maintenance sections of the MultiVOIP GSM software. How to update the firmware and software are also covered here should either be needed. This section will mainly focus on the Statistics section of the configuration software, but there are references to a few of the other sections as they are used more in the daily operations than in a setup situation.

Software Categories Covered in This Chapter

- **System Information**
- **Call Progress**
- **Wireless Statistics**
- **Endpoint Statistics**
- **Logs**
- **IP Statistics**
- **Link Management**
- **Registered Gateway Details**
- **Servers**
  - **SIP Proxies**
- **Advanced**
  - **Packetization Time**

System Information screen

This screen presents system information at a glance. It is found under the Configuration section and its primary use is in troubleshooting. The information presented in the figure below is for reference only and is not meant to be an exact match of your system.



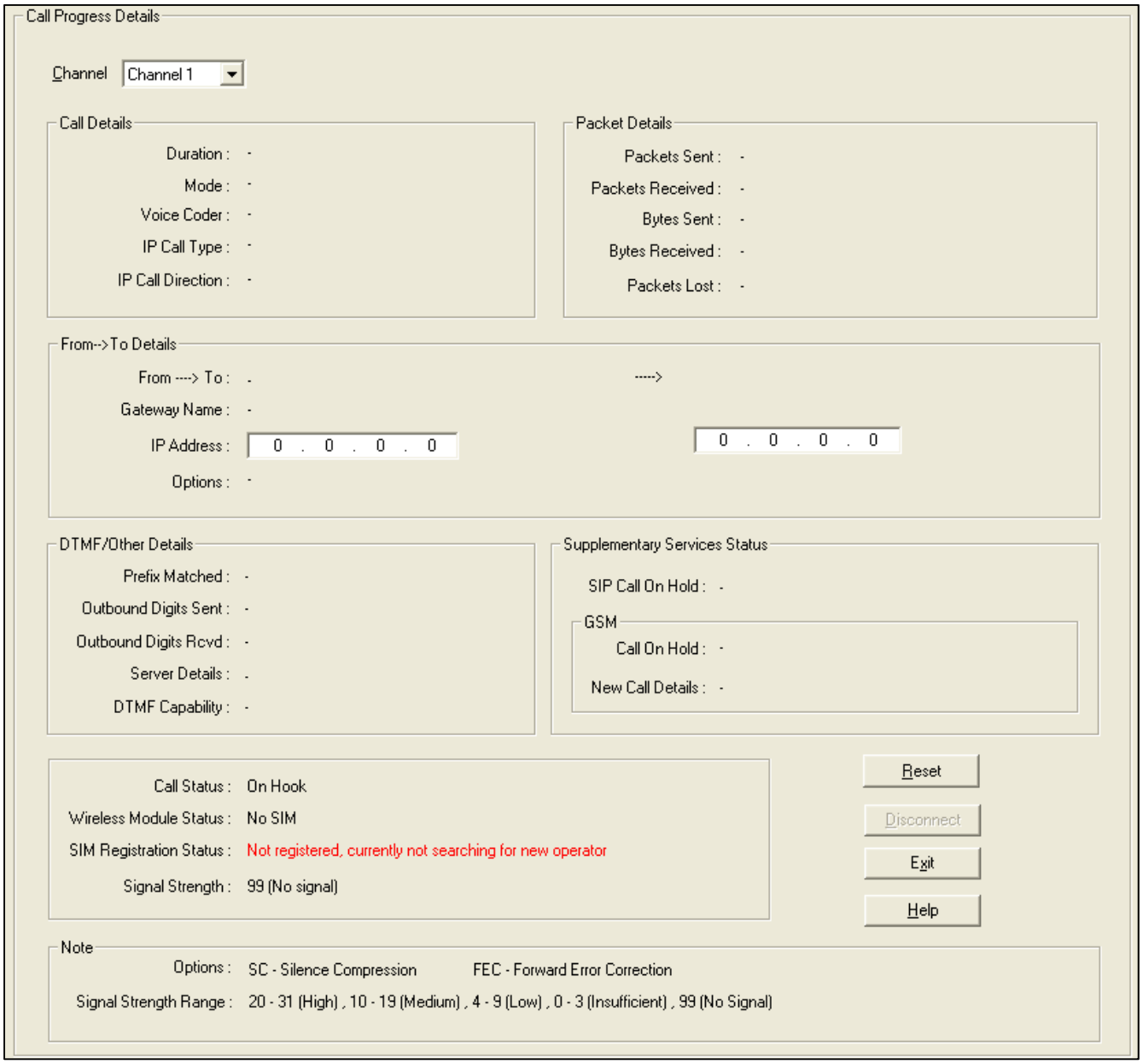
System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Version	*nn.nn* alpha-numeric	Indicates the version of the boot code that is running on the VOIP. The boot code version is independent of the software version.
Firmware Version	*nn.nn.nn* alpha-numeric	Indicates the version of the MultiVOIP GSM firmware.
Configuration Version	*nn.nn. nn.nn* alpha-numeric	Indicates the version of the MultiVOIP GSM configuration software.
Phone Book Version	*nn.nn* alpha-numeric	Indicates the version of the MultiVOIP GSM phone book being used.
Module *n*	Module Info	This will display the model information of the internal wireless modems.
Mac Address	numeric	Denotes the number assigned as the VOIP unit's unique MAC address.
Up Time	days: hours: mm:ss	Indicates how long the VOIP has been running since its last booting.
Hardware ID	alpha-numeric	Indicates version of the MultiVOIP GSM circuit board assembly being used.

The frequency with which the System Information screen is updated is determined by a setting in the Logs/Traces screen ("Online Statistics Update Interval").

Statistics Section

Ongoing operation of the MVPGSM, whether it is in a MVPGSM/PBX setting or MVPGSM/Telco-office setting, can be monitored for performance using the Statistics functions of the MVPGSM software. The following screens are examples of what can be shown and are followed by detailed descriptions of the categories involved. The model and signaling used will affect what is available for display.

Call Progress

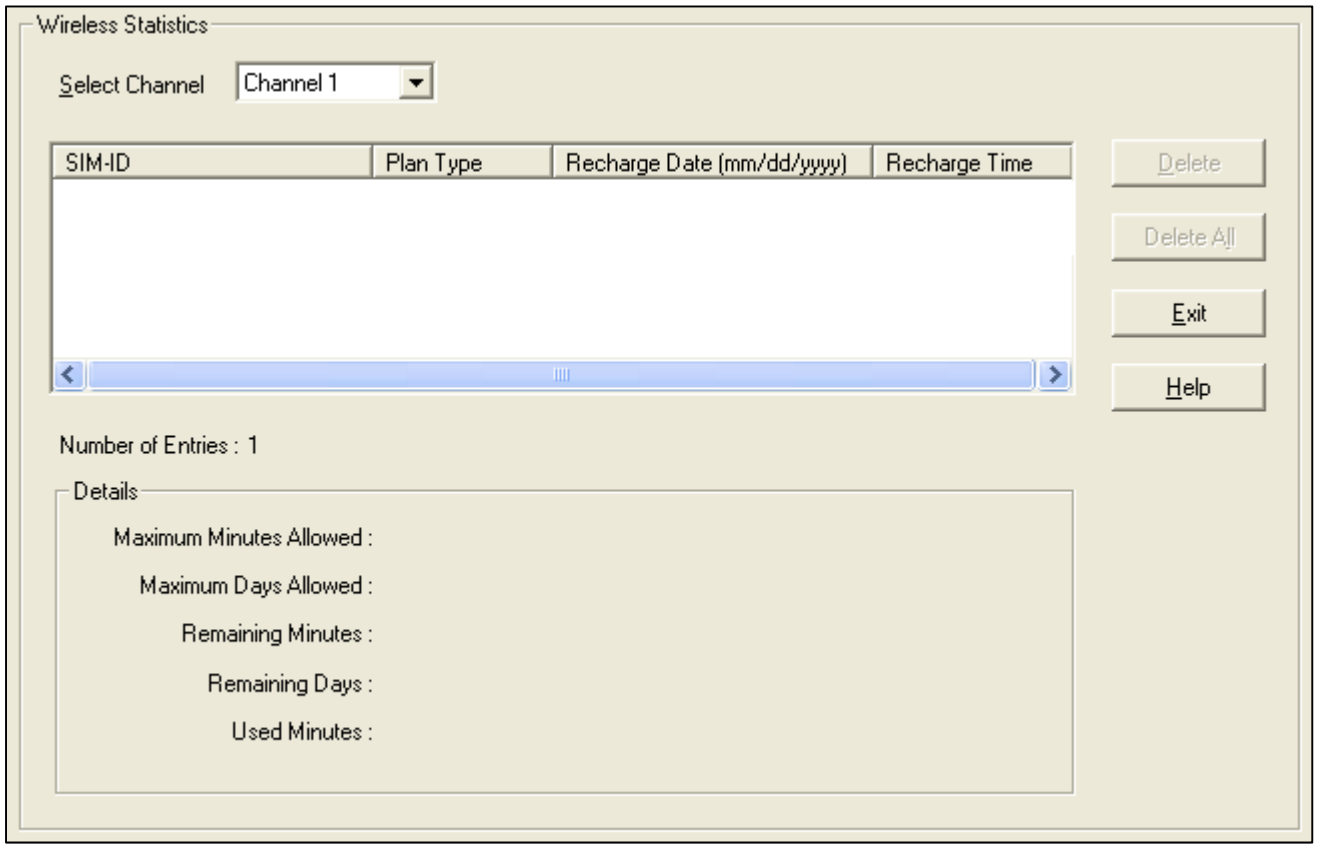


Call Progress Details: Field Definitions		
Field Name	**Values**	**Description**
Channel	1-n	This is the channel for which call-progress details are being viewed.
Call Details		
Duration	H/M/S	The length of the call in hours, minutes, and seconds (hh:mm:ss).
Mode	Voice or FAX	Indicates whether the call being described was a voice call or a FAX call.
Voice Coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.
IP Call Type	SIP	Indicates the Call Signaling protocol used for the call (SIP).
IP Call Direction	incoming, outgoing	Indicates whether the call in question is an incoming call or an outgoing call.
Packet Details		
Packets Sent	integer value	Number of voice packets sent over IP network in the course of this call.
Packets Rcvd	integer value	Number of voice packets received over IP network during this call.
Bytes Sent	integer value	Number of bytes sent over IP network in the course of this call.
Bytes Rcvd	integer value	Number of bytes received over IP network in the course of this call.
Packets Lost	integer value	Number of voice packets lost in the course of this call.
From – To Details		
Gateway Name (from)	alphanumeric string	Identifier for the SIP end point gateway that originated the call.
IP Address (from)	*n.n.n.n*	IP address for the SIP end point that originated the call.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.
Gateway Name (to)	alphanumeric	Identifier for the VOIP gateway that received the call.
IP Address (to)	*n.n.n.n*	IP address for the VOIP gateway that received the call.
Options	SC, FEC	Displays VOIP transmission options in use on the current call
DTMF/Other Details		
Prefix Matched	specified dialing digits	Displays the dialed digits that were matched to a phonebook entry.
Outbound Digits Sent	0-9, #, *	The digits transmitted by the MultiVOIP GSM for this call.
Outbound Digits Received	0-9, #, *	Of the digits transmitted by the MultiVOIP GSM for this call; these are the DTMF digits that are received from the IP network side.
Server Details	*n.n.n.n* and/or other related descriptions	The IP address (etc.) of the traffic control server (if any) being used (SIP proxy) will be displayed here if the call is handled through that server.
DTMF Capability	inband, out of band Expressions differ slightly for different Call Signaling protocols.	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition.

Table is continued on next page…

Call Progress Details: Field Definitions (continued)		
Field Name	**Values**	**Description**
Supplementary Services Status		
SIP Call on Hold	alphanumeric	Describes held call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers come from **Gateway Name** field in **Phone Book Configuration** screen of remote VOIP.
GSM Call on Hold	alphanumeric	Displays the phone number/SIP URL of a call that is on hold.
GSM New Call Details	alphanumeric	Displays the phone number/SIP URL of a newly generated call.
Call Status fields		
Call Status	OnHook; Call Setup; Ringback; Active; Hangup	Shows condition of current call.
Wireless Module Status	Not Responding; Disabled; No SIM; Ready; SIM PIN Required; SIM PUK Required	This displays the current status of the wireless module on this channel.
SIM Registration Status	"Not registered, currently not searching for a new operator"; "Registered to home network"; "Not registered, currently searching for a new operator"; "Registration denied"; "Unknown"; "Registered, roaming"	This shows the registration status of the module on this channel.
Signal Strength	0-99	0-3 is an insufficient signal for use. 4-9 is a low strength signal that is usable. 10-19 is a medium strength and is a good connection. 20-31 is a high strength and is an excellent connection. 99 means there is no signal at all.
Note section (Options & Signal Strength Ranges)		"SC" stands for Silence Compression. With **Silence Compression** enabled, the MultiVOIP GSM will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. "FEC" stands for Forward Error Correction. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel.
Reset (button)	-	Clicking this button will reset the channel including the wireless module.
Disconnect (button)	-	Clicking this button will disconnect the current connection on this channel.

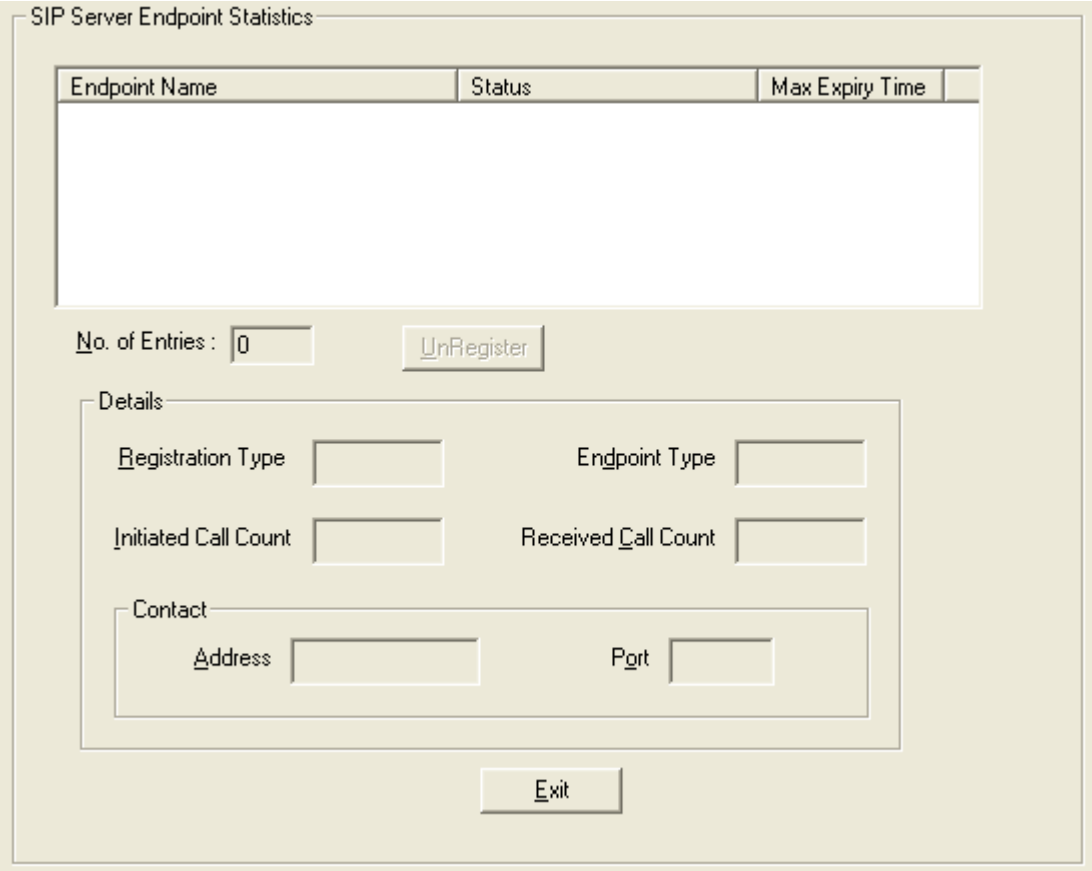
Wireless Statistics



Wireless Statistics parameters		
Field Name	**Values**	**Description**
Select Channel	1 – 2	Select the channel that you want to see statistics for.
SIM-ID	number	This will display the ID number of the SIM card on the selected channel.
Plan Type*	SIM plan	This will display the plan that the SIM card has.
Recharge Date*	(mm/dd/yyyy)	This will display the date (month/day/year format) when a plan needs to be recharged (if necessary).
Recharge Time*	time	This will display the time **12:00 A.M.** when a plan needs to be recharged.
Delete	(button)	This will permanently delete the selected statistic line.
Delete All	(button)	This will permanently delete all statistic lines.
Number of Entries	number	This simply shows the total number of statistical entries present.
Details	varies	This will display information for the currently selected SIM-ID.

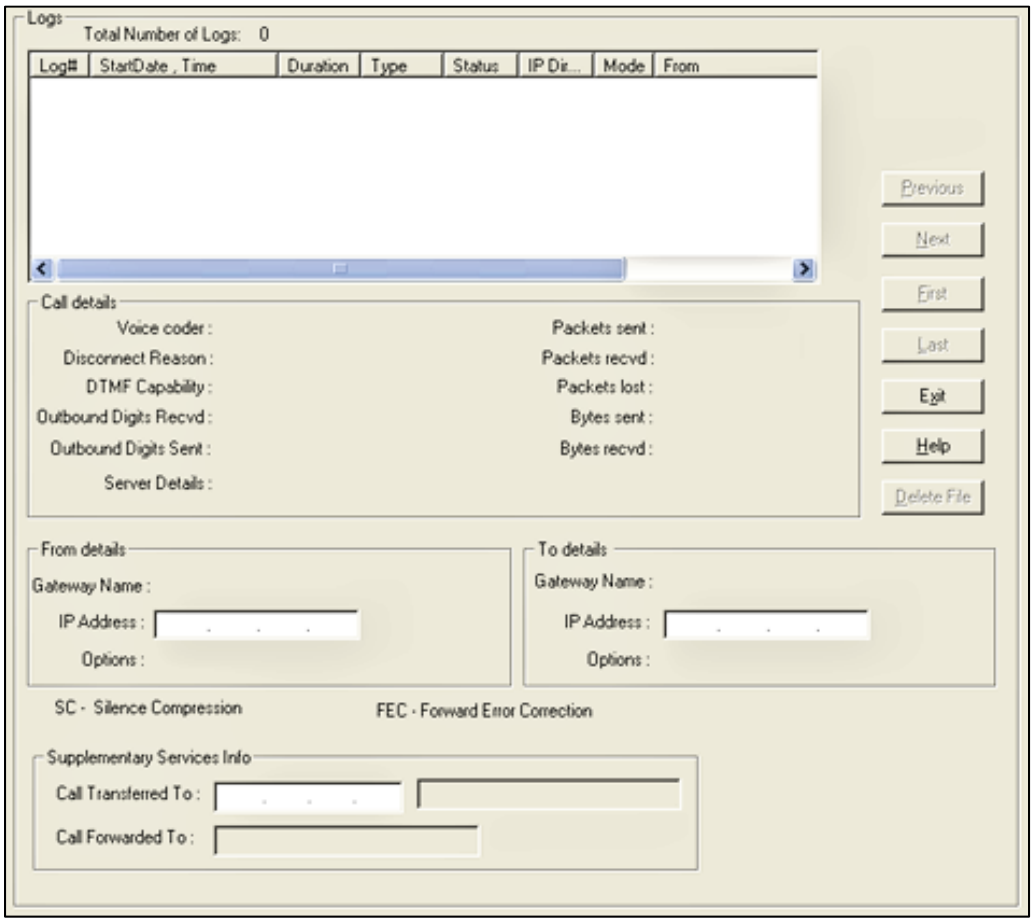
* These items are based on the user configured Calling Plan screen in the Wireless Setup section of Configuration.

Endpoint Statistics



SIP Server Endpoint Statistics		
Endpoint Name	list column	Displays the name of the endpoint.
Status	list column	Shows the status of the endpoint.
Max Expiry Time	list column	Displays the maximum set time to expire.
No. of Entries	number	Displays the total number of entries in the section above.
Unregister	button	Click this button to unregister the endpoint that is selected from the list above.
Details		
Registration Type		
Endpoint Type		
Initiated Call Count	number	Total number of calls initiated will be displayed for the selected endpoint.
Received Call Count	number	Total number of calls received will be displayed for the selected endpoint.
Contact		
Address	n.n.n.n	This will show the IP address for the selected endpoint.
Port	number	This will show the port for the selected endpoint.

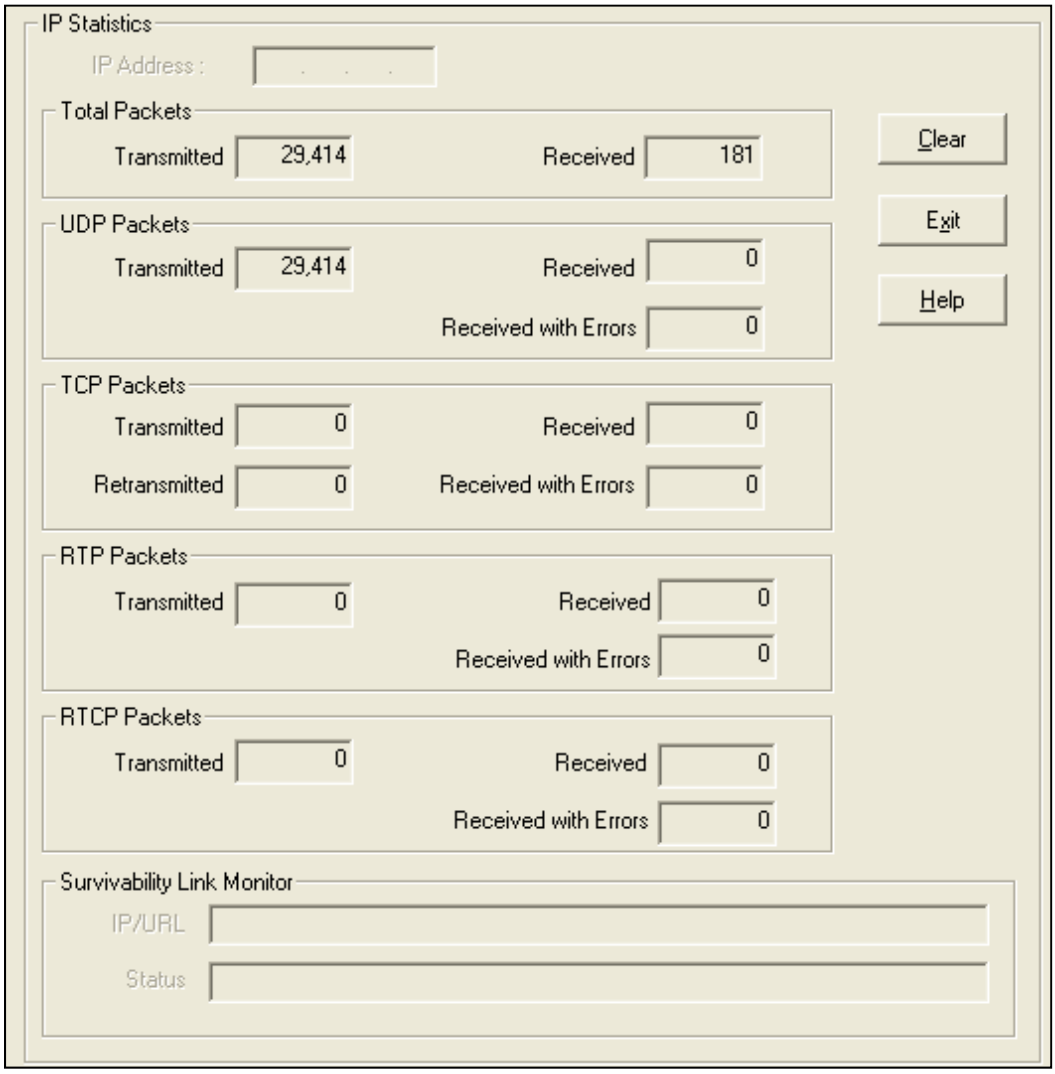
Logs



The table below describes the fields of the Logs screen.

Logs Screen Details: Field Definitions		
Field Name	**Values**	**Description**
Log # column	1 or higher	All calls are assigned an event number in chronological order, with the most recent call having the highest event number.
Start Date, Time column	dd:mm:yyyy hh:mm:ss	The starting time of the call. The date is presented as a day and a month of one or two digits, and a four-digit year, followed by a time-of-day.
Duration column	hh:mm:ss	This describes how long the call lasted in hours, minutes, and seconds.
Type	SIP	Indicates the Call Signaling protocol used for the call (SIP).
Status column	success or failure	Displays the status of the call (whether the call was completed or not).
IP Direction	Incoming, outgoing	Shows call "incoming" or "outgoing" with respect to the gateway.
Mode column	voice or FAX	Shows if the event being described was a voice call or a FAX call.
From column	gateway name	Displays the name of the voice gateway that originates the call.
To column	gateway name	Displays the name of the voice gateway that completes the call.
Special Buttons		
Previous	--	Displays log entry before currently selected one.
Next	--	Displays log entry after currently selected one.
First	--	Displays first log entry
Last	--	Displays last log entry.
Delete File	--	Deletes selected log file.
Call Details		
Voice coder	Coder protocol	The voice coder being used on this call.
Disconnect Reason	"Normal" or "Local"	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause.
DTMF Capability	inband, out of band	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different VOIP protocols. For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition.
Outbound Digits Received	0-9, #, *	These are the digits received from the remote SIP end point that are used to initiate a call out on the wireless network.
Outbound Digits Sent	0-9, #, *	These are the digits received from the wireless network and sent to the remote SIP end point to place a call.
Server Details	*n.n.n.n*	When the MultiVOIP GSM is operating in the non-direct mode (with proxy in SIP mode), this field shows the IP address of the server that is directing IP phone traffic.
Packets sent	integer value	Number of voice packets sent over the IP network during this call.
Packets received	integer value	Number of voice packets received over the IP network in the course of this call.
Packets lost	integer value	Number of voice packets from this call that were lost after being received from the IP network.
Bytes sent	integer value	Number of bytes of voice sent over the IP network during this call.
Bytes received	integer value	Number of bytes of voice received over the IP network during this call.
FROM Details		
Gateway Name	alphanumeric	Identifier for the VOIP gateway that originated this call.
IP Address	*n.n.n.n*	IP address for the VOIP gateway that originated this call.
Options	FEC, SC	Displays transmission options used by the gateway originating the call.
TO Details		
Gateway Name	alphanumeric	Identifier for the VOIP gateway that received this call.
IP Address	*n.n.n.n*	IP address for the VOIP gateway that received this call.
Options		Displays transmission options used by VOIP gateway receiving the call.
Supplementary Services Info		
Call Transferred To	phone number	Number of party called in transfer.
Call Forwarded To	phone number	Number of party called in forwarding.

IP Statistics



UDP versus TCP (User Datagram Protocol versus Transmission Control Protocol)

UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission of data.

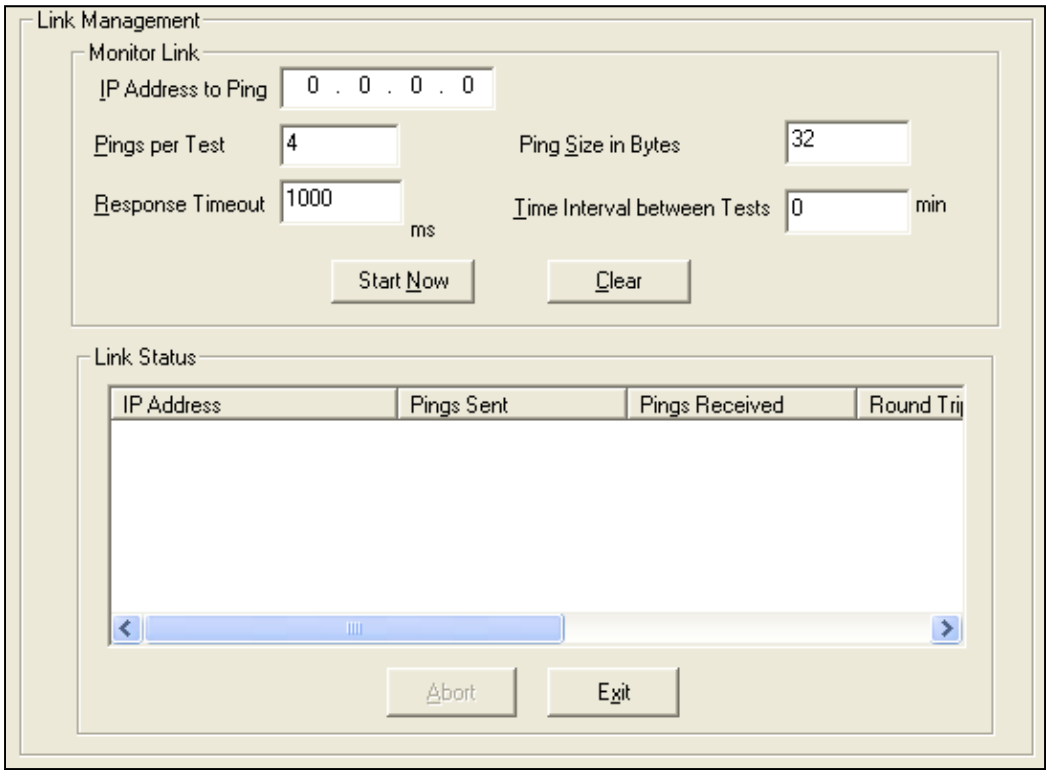
Both TCP and UDP split data into packets called “datagrams.” However, TCP includes extra headers in the datagram to enable retransmission of lost packets and reassembly of packets into their correct order if they arrive out of order. UDP does not provide this. Lost UDP packets are irretrievable; that is, out-of-order UDP packets cannot be reconstituted in their proper order.

Despite these obvious disadvantages, UDP packets can be transmitted much faster than TCP packets -- as much as three times faster. In certain applications, like audio and video data transmission, the need for high speed outweighs the need for verified data integrity. Sound or pictures often remain intelligible despite a certain amount of lost or disordered data packets (which comes through as static).

IP Statistics: Field Definitions		
Field Name	**Values**	**Description**
IP Address	*n.n.n.n*	IP address of the MultiVOIP GSM. For an IP address to be displayed here, the MultiVOIP GSM must have DHCP enabled. Its IP address, in such a case, is assigned by the DHCP server.
"Clear"	button	Clears packet tallies from memory.
Total Packets		Sum of packets of all types.
Transmitted	integer value	Total number of packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received	integer value	Total number of packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received with Errors	integer value	Total number of error-laden packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
UDP Packets		User Datagram Protocol packets.
Transmitted	integer value	Number of UDP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received	integer value	Number of UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received with Errors	integer value	Number of error-laden UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
TCP Packets		Transmission Control Protocol packets.
Transmitted	integer value	Number of TCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received	integer value	Number of TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received with Errors	integer value	Number of error-laden TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
RTP Packets		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received	integer value	Number of RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received with Errors	integer value	Number of error-laden RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
RTCP Packets		Realtime Transport Control Protocol packets convey control information to assist in the transmission of RTP (voice) packets. RTCP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received	integer value	Number of RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Received with Errors	integer value	Number of error-laden RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM software.
Survivability Link Monitor		
IP/URL	n.n.n.n	Displays the current IP address being used to PING or the Internet address that is being used for attempted connections for local survivability.
Status	text	This will display the status of the last survivability result. Status values: *Trying: DNS Request*; *Trying: Ping Request*; *Trying: TCP Connect*; *Link Up: Ping Response Received*; *Link Up: TCP Connect Success*; *Link Down: DNS – Failed to Resolve URL*; *Link Down: DNS Times Out*; *Link Down: Ping Request Timed Out*; *Link Down: Not Reachable*; *Link Down: TCP Connect Timed Out*

Link Management

The Link Management screen is essentially an automated utility for pinging endpoints on your VOIP network. This utility generates pings of variable sizes at variable intervals and records the response to the pings.

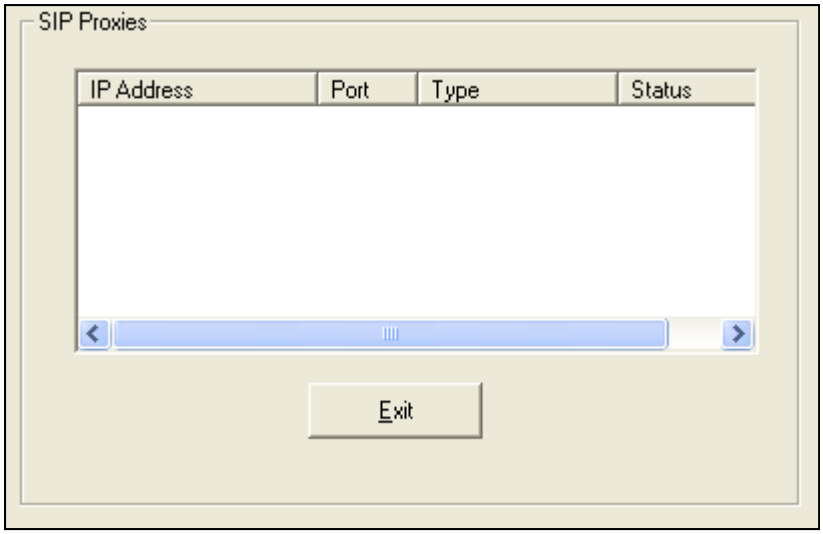


Link Management screen Field Definitions		
Field Name	**Values**	**Description**
Monitor Link fields		
IP Address to Ping	*n.n.n.n*	This is the IP address of the target endpoint to be pinged.
Pings per Test	1-999	This field determines how many pings will be generated by the Start Now command.
Response Timeout	500 – 5000 milliseconds	The duration after which a ping will be considered to have failed.
Ping Size in Bytes	32 – 128 bytes	This field determines how long or large the ping will be.
Timer Interval between Pings	0 or 30 – 6000 minutes	This field determines how long of a wait there is between one ping and the next.
Start Now button	--	Initiates pinging.
Clear button	--	Erases ping parameters in Monitor Link field group and restores default values.
Link Status Parameters		These fields summarize the results of pinging.
IP Address column	*n.n.n.n*	Target of ping.
No. of Pings Sent	as listed	Number of pings sent to target endpoint.
No. of Pings Received	as listed	Number of pings received by target endpoint.
Round Trip Delay (Min/Max/Avg)	as listed, in milliseconds	Displays how long it took from time ping was sent to time ping response was received.
Last Error	as listed	Indicates when last data error occurred.

Servers

SIP Proxies

This window lists all the SIP proxy servers configured and the active server with which the system is registered.

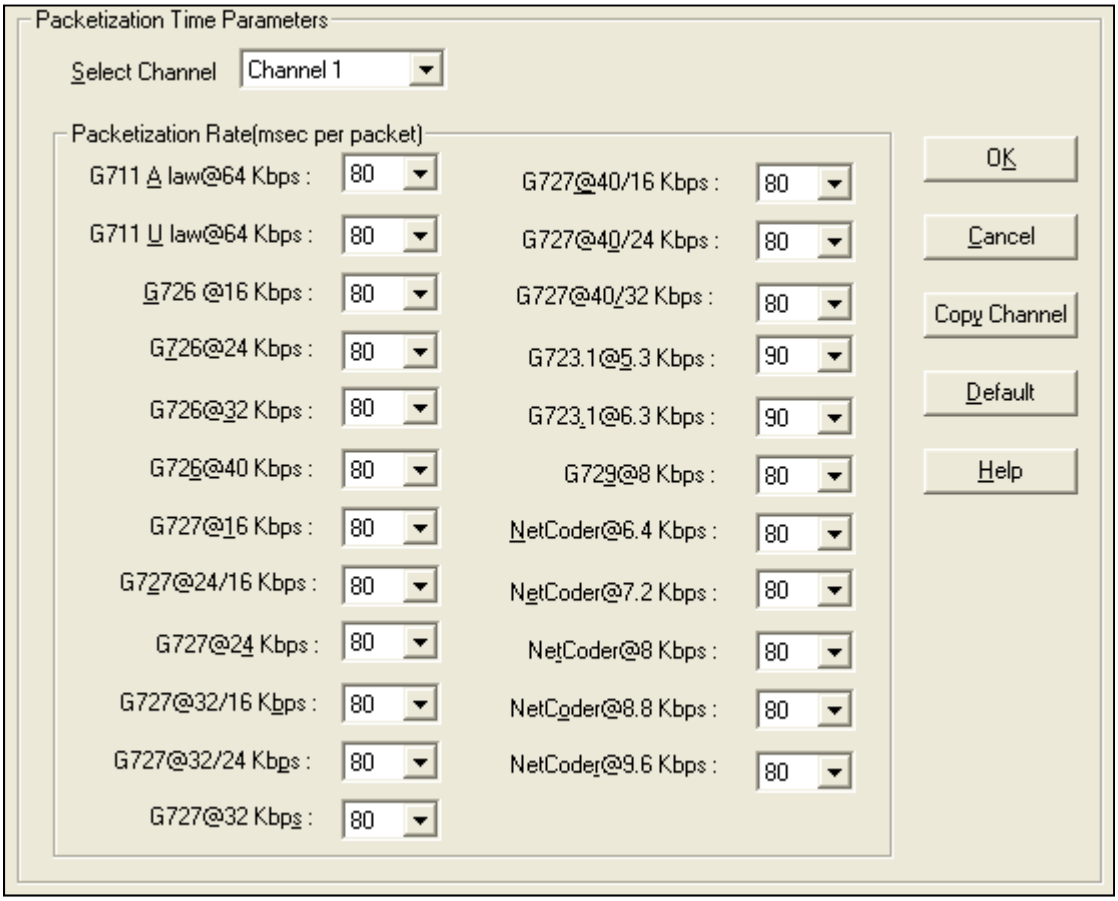


SIP Proxies (Statistics, Servers): Field Definitions		
Field Name	**Values**	**Description**
IP Address	*n.n.n.n*	The IP address of the SIP proxy by which the MultiVOIP GSM is governed.
Port	*port*	SIP signaling port used for communication between MultiVOIP GSM unit and the SIP Proxy that governs it.
Type	Primary, Alternate	This field describes the type of gateway the MultiVOIP GSM is defined as with respect to the SIP proxy server.
Status	registered, not registered	The current status of the MultiVOIP GSM gateway with respect to the SIP proxy either registered or unregistered.

Advanced

Packetization Time

You can use the **Packetization Time** screen to specify definite packetization rates for coders selected in the Voice/FAX Parameters screen (in the "Coder Options" group of fields). The Packetization Time screen is accessible under the "Advanced" options entry in the sidebar list of the main VOIP software screen. In dealing with RTP parameters, the Packetization Time screen is closely related to both Voice/FAX Parameters and to IP Statistics. It is located in the "Advanced" group for ease of use.



Packetization rates can be set separately for each channel.

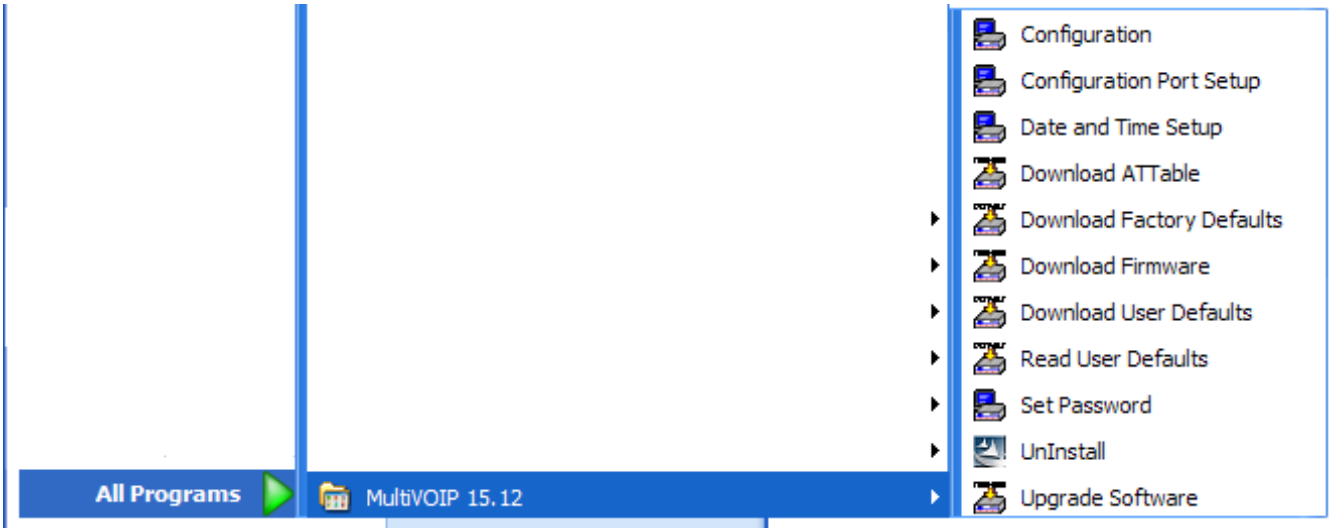
The table below presents the ranges and increments for packetization rates. The final column represents recommended settings (based on the most common found) when operating with third party devices.

Packetization Ranges and Increments				**Recommendations**
Coder Types	Range (in Kbps); {default}		Increments (in Kbps)	Setting (in ms)
G711, G726, G727	5-120	{5}	5	20
G723	30-120	{30}	30	30
G729	10-120	{10}	10	20
NetCoder	20-120	{20}	20	20

Once the packetization rate has been set for one channel, it can be copied into the other channel by using the Copy Channel button on the Packetization Time screen.

MultiVOIP GSM Program Menu Items

After the MultiVOIP GSM program is installed on the PC, it can be launched from the **Programs** group of the Windows **Start** menu (**Start** | **Programs** | **MultiVOIP GSM x.xx** | …). In this section, we describe the software functions available on this menu.



Several basic software functions are accessible from the MultiVOIP GSM software menu, as shown below.

MultiVOIP GSM Program Menu	
Menu Selection	**Description**
Configuration	Select this to enter the Configuration program where values for IP, telephony, and other parameters are set.
Configuration Port Setup	Select this to access the **COM Port Setup** screen of the MultiVOIP GSM Configuration program.
Date and Time Setup	Select this for access to set calendar/clock used for call logging.
Download ATTable	Select this to download the ATTable files. This is generally only needed in conjunction with a firmware update.
Download Factory Defaults	Select this to return the configuration parameters to the original factory values.
Download Firmware	Select this to initiate a firmware download.
Download User Defaults	To be used after a full set of parameter values, values specified by the user, have been saved (using Save Setup). This command loads the saved user defaults into the MultiVOIP GSM.
Read User Defaults	This will read the User defaults currently set to the MVPGSM unit and save them to the attached computer. The areas read are: Configuration. Inbound/Outbound Phone Directories, Tariff Plan, Load Balancing Groups, Local Survivability and the CLIP Routing database.
Set Password	Select this to create a password for access to the MultiVOIP GSM software programs (**Program** group commands, Windows interface, web browser interface, & FTP server). Only the FTP Server function *requires* a password for access. The FTP Server function also requires that a username be set along with the password.
Uninstall	Select this to uninstall the MultiVOIP GSM software (most, but not all components are removed from computer when this command is used).
Upgrade Software	Loads firmware and settings from the controller PC to the MultiVOIP GSM unit. User can choose whether to load Factory Default Settings or Current Configuration settings.

"Downloading" here refers to transferring program files from the PC to the nonvolatile "flash" memory of the MultiVOIP GSM. Such transfers are made via the PC's serial port. This can be understood as a "download" from the perspective of the MultiVOIP GSM unit.

When new versions of the MultiVOIP GSM software become available, they will be posted on Multi-Tech's website. Although transferring updated program files from the Multi-Tech website to the user's PC can generally be considered a download (from the perspective of the PC), this type of download cannot be initiated from the MultiVOIP GSM software's Program menu command set.

Setting and Downloading User Defaults

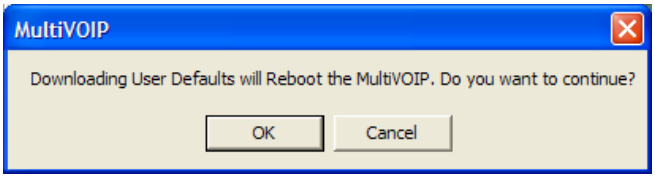
The **Download User Defaults** command allows you to maintain a known working configuration that is specific to your VOIP system. You can then experiment with alterations or improvements to the configurations confident that a working configuration can be restored if necessary.

1. Before you can use the Download User Defaults command, you must first save a set of configuration parameters by using the **Save Setup** command in the sidebar menu of the MultiVOIP GSM software.

2. Before the setup configuration is saved, you will be prompted to save the setup as the User Default Configuration. Select the checkbox and click **OK**.

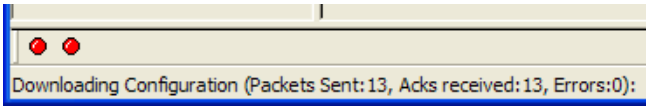
   A user default file will be created. The MultiVOIP GSM unit will reboot itself.

3. To download the user defaults, go to **Start | Programs | MultiVOIP GSM x.xx | Download User Defaults**.
4. A confirmation screen will appear indicating that this action will entail rebooting the MultiVOIP GSM.

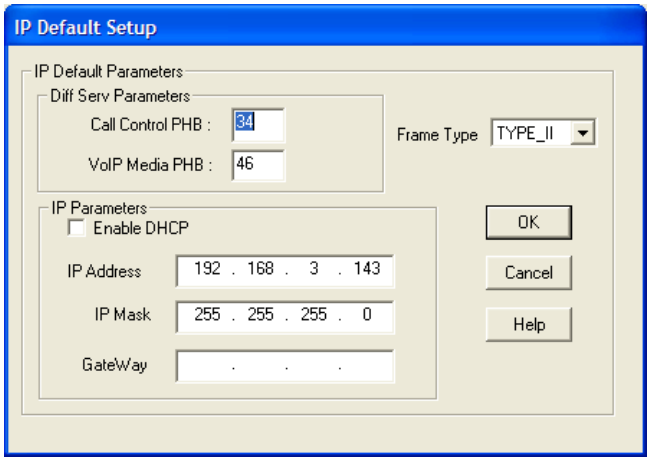


   Click **OK**.

5. Progress bars will appear during the file transfer process.



6. When the file transfer process is complete, the **Dialog / IP Parameters** screen will appear.



7. Set the IP values per your particular VOIP system. Click **OK**. Progress bars will appear as the MultiVOIP GSM reboots itself.

Setting a Password

Windows Interface

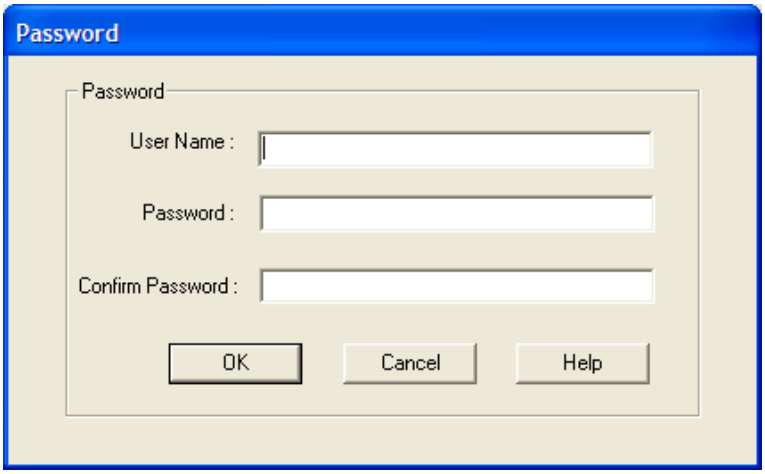
After a user name has been designated and a password has been set, that password is required to gain access to any functionality of the MultiVOIP GSM software. Only one user name and password can be assigned to a VOIP unit. The user name will be required when communicating with the MultiVOIP GSM via the web browser interface.

NOTE: Record your user name and password in a safe place. If the password is lost, forgotten, or irretrievable, the instructions on the next page, Password Reset, will allow you to reset the password to the factory default.

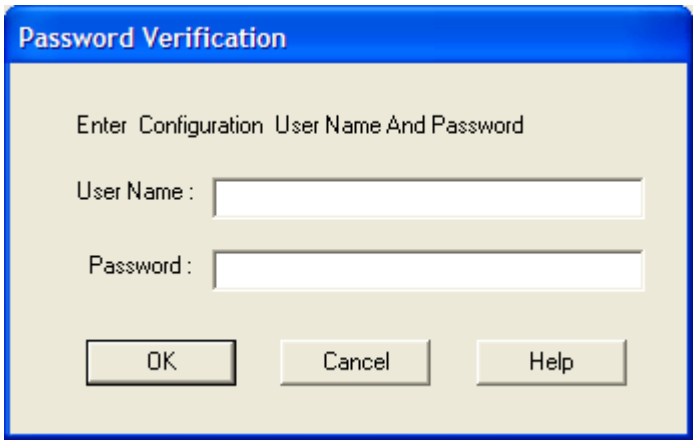
1. The MultiVOIP GSM configuration program must be off when invoking the **Set Password** command. If it is on, the command will not work.
2. To use the **Set Password** command, go to **Start** | **Programs** | **MultiVOIP GSM x.xx** | **Set Password**.
3. You will be prompted to confirm that you want to establish a password, which will entail rebooting the MultiVOIP GSM (which is done automatically).

   Click **OK** to proceed with establishing a password.
4. The **Password** screen will appear. If you intend to use the FTP Server function that is built into the MultiVOIP GSM, enter a user name. (A User Name is not needed to access the local Windows interface, the web browser interface, or the commands in the **Program** group.) Type your password in the **Password** field of the **Password** screen. Type this same password again in the **Confirm Password** field to verify the password you have chosen.

   NOTE: Be sure to write down your password in a convenient but secure place.



   Click **OK**.
5. A message will appear indicating that a password has been set successfully.
   After the password has been set successfully, the MultiVOIP GSM will re-boot itself and, in so doing, its **BOOT** LED will light up.
6. After the password has been set, the user will be required to enter the password to gain access to the web browser interface and any part of the MultiVOIP GSM software listed in the **Program** group menu. User Name and Password are both needed for access to the FTP Server residing in the MultiVOIP GSM.



When the MVPGSM program asks for the password at launch of program, if **CANCEL** is selected, the program will simply shut down.

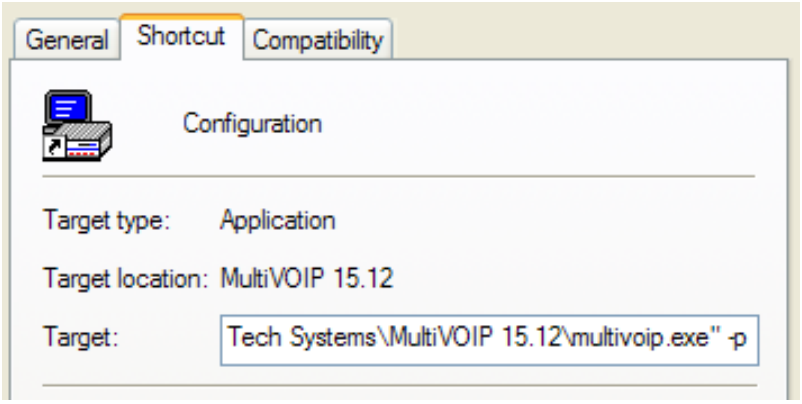
The MVPGSM program will produce an error message if an invalid password is entered.

Password Reset

1. Connect your PC serial COM port to the command port on the back of the MultiVOIP unit.
2. Open HyperTerminal or other communications software and set the software COM port to match the settings of the COM port that is attached to the MultiVOIP. [The MVPGSM only has two settings for port speed: 115200 (default) and 19200. Ensure your COM port is using one of these baud rates.]
3. Power cycle (power off and then back on) the MultiVOIP. You should be able to see some messages printing out on the HyperTerminal (or other software) screen. If you are not seeing any messages, it means you either do not have the HyperTerminal (or other) software set to match the comport correctly or you have a bad cable.
4. The last message on the HyperTerminal (or other software) screen will tell you what the firmware version of the MultiVOIP is. Write it down as you will need to know this for the next step.
5. Then go to our Internet site at ftp://ftp.multitech.com/multivoip and download the software version corresponding to the firmware version that you have written down from step 4.
   For example, if your firmware version is listed in the system information as "Firmware version: 15.11.34" - it means you need to download the file named: *mvp151134.exe* from our FTP site.
6. Unzip the software and install it on the PC.
7. Click on the Start button and navigate to Programs -> MultiVOIP *xx.xx* -> Configuration. Left-click this selection to set up your comport for the console connection, if needed. Click OK when you are done.
   Note: *xx.xx* refers to the software version currently installed.
8. Click on the Start button and navigate to Programs and highlight (but do not left-click) *MultiVOIP xx.xx*. Right click on the highlighted selection (*MultiVOIP xx.xx*) and select 'Open.' This will bring up the program group for editing.
9. Highlight the *Configuration* shortcut icon and right-click on it (do not double left-click to launch the configuration program) and select the 'Properties' option at the bottom. We need to alter the target settings.

10. The shortcut tab should be opened by default. In the 'Target:' text box, put a space, a minus sign and the letter p after the last double quotation mark.

    Example: "C:\Program Files\Multi-Tech Systems\MultiVOIP 15.11\multivoip.exe" ***–p***



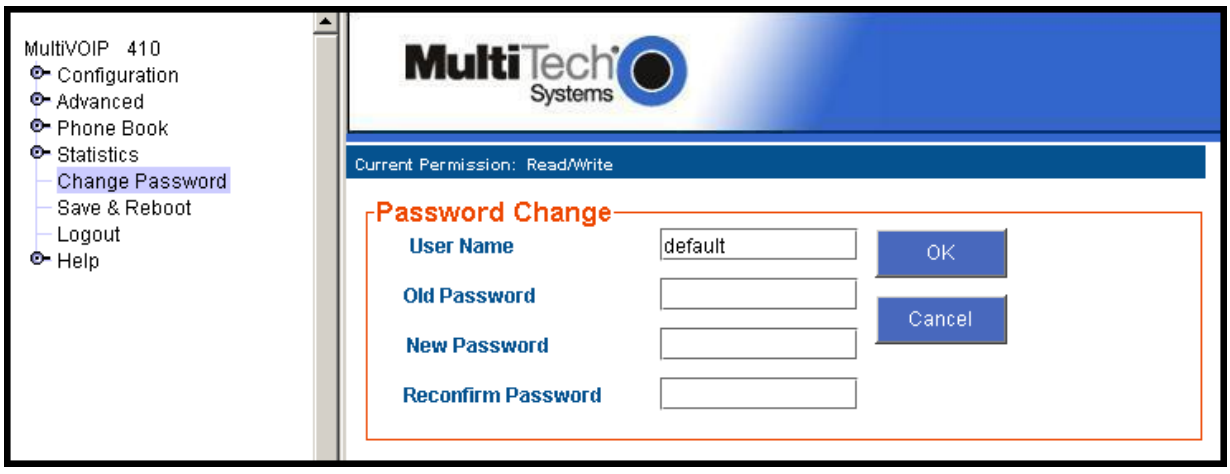
    Click OK.

11. Now double click on the Configuration shortcut icon. It will ask for confirmation on resetting the password. Click OK.
12. It will now asking for new username and password. Enter your desired username and password. Click OK to save it.
13. Now repeat steps **9** and **10** to **remove** the '***-p***' that was added to return the Configuration link to its normal state.
14. You are done.

Web Browser Interface

Setting a password is optional when using the MultiVOIP GSM web browser interface. Only one password can be assigned and it works for all MultiVOIP GSM software functions (Windows interface, web browser interface, FTP server, and all Program menu commands, *e.g.*, Upgrade Software – only the FTP Server function requires a User Name in addition to the password). After a password has been set, that password is required to access the MultiVOIP GSM web browser interface.

NOTE: Record your user name and password in a safe place.



Upgrading Software

As noted earlier, the Upgrade Software command transfers from the controller PC to the MultiVOIP GSM unit. The settings can be either Factory Default Settings or Current Configuration Settings.

FTP Server File Transfers ("Downloads")

Multi-Tech has built an FTP server into the MultiVOIP GSM unit. Therefore, file transfers from the controller PC to the VOIP unit can be done using an FTP client program or even using a browser (e.g., Internet Explorer, Netscape, or Firefox, used in conjunction with Windows Explorer).

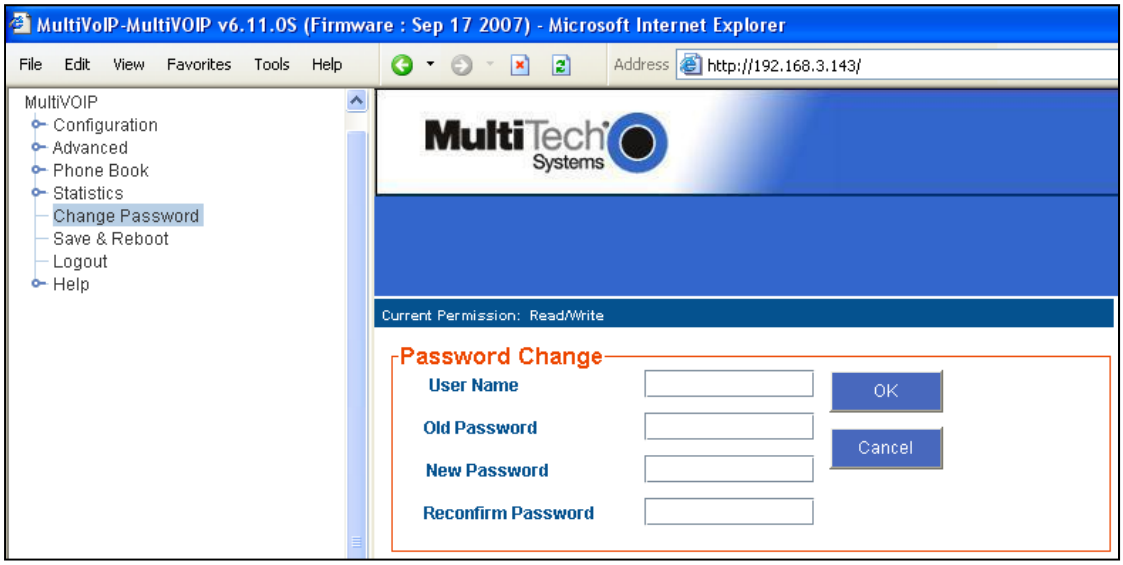
The terminology of "downloads" and "uploads" gets a bit confusing in this context. File transfers from a client to a server are typically considered "uploads." File transfers from a large repository of data to machines with less data capacity are considered "downloads." In this case, these metaphors are contradictory: the FTP server is actually housed in the MultiVOIP GSM unit, and the controller PC, which is actually the repository of the info to be transferred, uses an FTP client program. In this situation, we have chosen to call the transfer of files from the PC to the VOIP "downloads." (Be aware that some FTP client programs may use the opposite terminology, i.e., they may refer to the file transfer as an "upload ")

You can download firmware, default configuration parameters, and phonebook data for the MultiVOIP GSM unit with this FTP functionality. These downloads are done over a network, not by a local serial port connection. Consequently, VOIPs at distant locations can be updated from a central control point.

The phonebook downloading feature greatly reduces the data-entry required to establish inbound and outbound phonebooks for the VOIP units within a system. Although each MultiVOIP GSM unit will require some unique phonebook entries, most will be common to the entire VOIP system. After the phonebooks for the first few VOIP units have been compiled, phonebooks for additional VOIPs become much simpler: you copy the common material by downloading and then do data entry for the few phonebook items that are unique to that particular VOIP unit or VOIP site.

To transfer files using the FTP server functionality in the MultiVOIP GSM, follow these directions.

1. **Establish Network Connection and IP Addresses**. Both the controller PC and the MultiVOIP GSM unit(s) must be connected to the same IP network. An IP address must be assigned for each.
2. **Establish User Name and Password**. You must establish a user name and (optionally) a password for contacting the VOIP over the IP network. (When connection is made via a local serial connection between the PC and the VOIP unit, no user name is needed.)

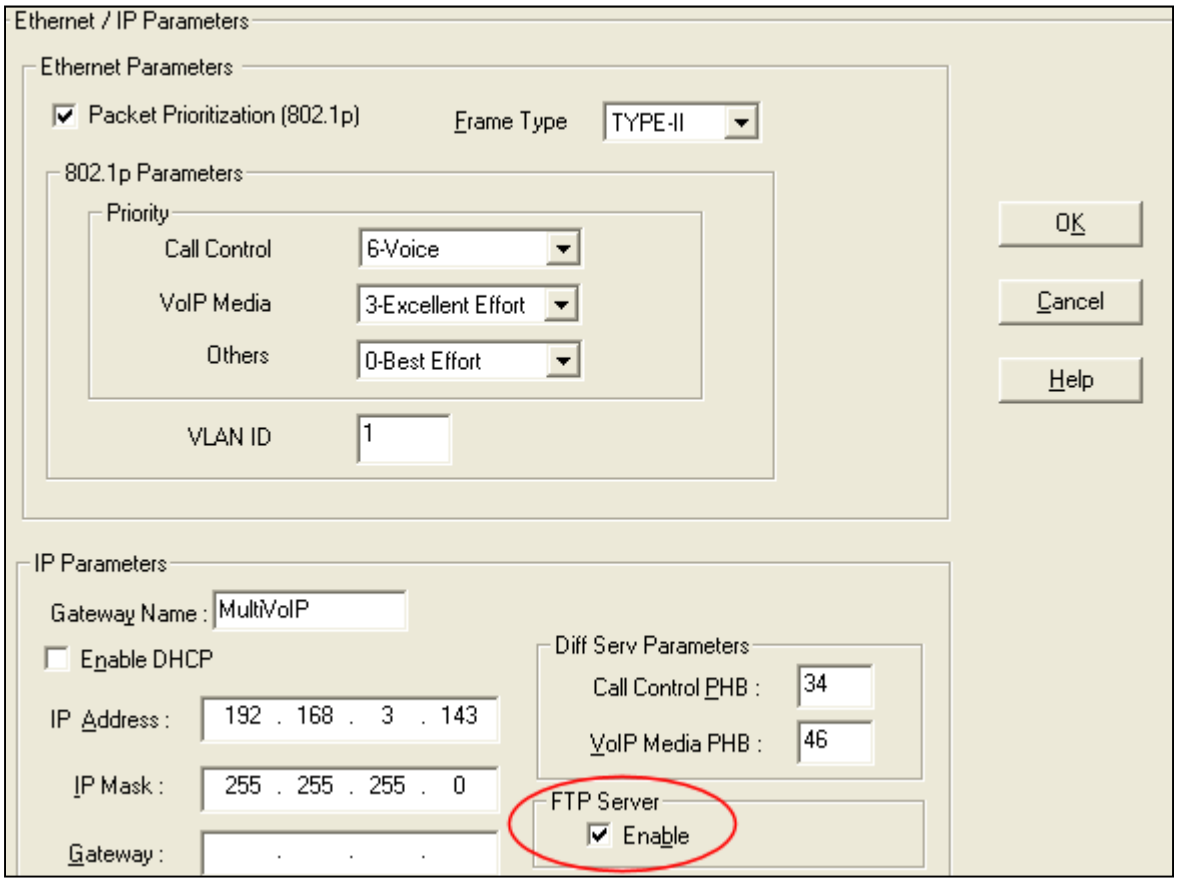


As shown above, the user name and password can be set in the web interface as well as in the Windows interface.

3. **Install FTP Client Program or Use Substitute**. You *should* install an FTP client program on the controller PC. FTP file transfers can be done using a web browser (e.g., Firefox or Internet Explorer) in conjunction with a local Windows browser a (e.g., Windows Explorer), but this approach is somewhat clumsy (it requires use of two application programs rather than one) and it limits downloading to only one VOIP unit at a time. With an FTP client program, multiple VOIPs can receive FTP file transmissions in response to a single command (the transfers may occur serially however).

   Although Multi-Tech does not provide an FTP client program with the MultiVOIP GSM software or endorse any particular FTP client program, we remind our readers that adequate FTP programs are readily available under retail, shareware and freeware licenses. (Read and observe any End-User License Agreement carefully.) Two examples of this are the "WSFTP" client and the "SmartFTP" client, with the former having an essentially text-based interface and the latter having a more graphically oriented interface, as of this writing. User preferences will vary.

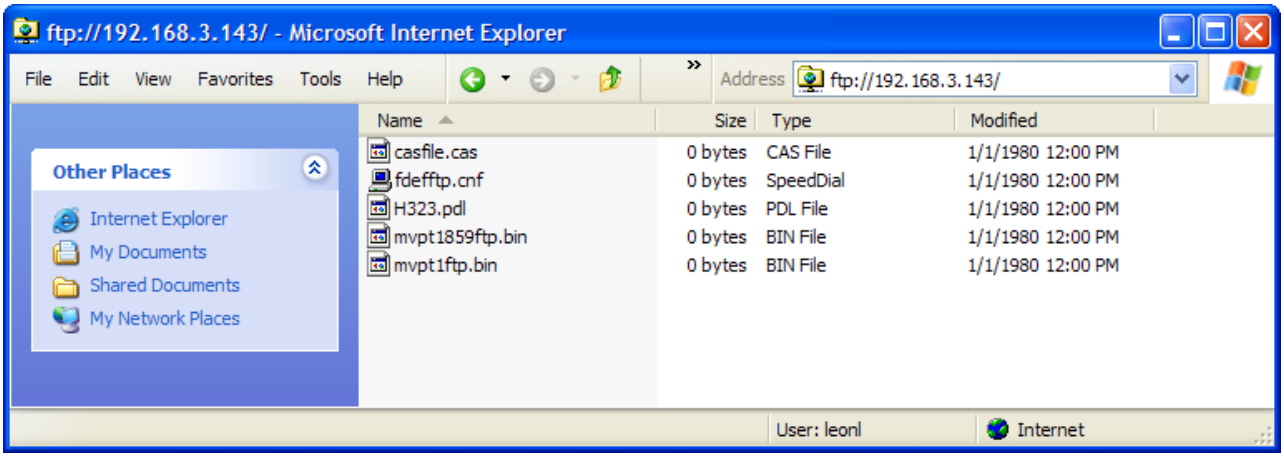
4. **Enable FTP Functionality**. Go to the **IP Parameters** screen and click on the "FTP Server: Enable" box.



5. **Identify Files to be Updated**. Determine which files you want to update. See table below.

File Type	File Names	Description
factory defaults	fdefFtp.cnf	This file contains factory default settings for user-changeable configuration parameters. Only one file of this type will be in the directory.
inbound phonebook	InPhBk.tmr	This file updates the inbound phonebook in the MultiVOIP GSM unit.
outbound phonebook	OutPhBk.tmr	This file updates the outbound phonebook in the MultiVOIP GSM unit.

6. **Contact MultiVOIP GSM FTP Server**. You must make contact with the FTP Server in the VOIP using either a web browser or FTP client program. Enter the IP address of the MultiVOIP GSM's FTP Server. If you are using a browser, the address must be preceded by "ftp://" (otherwise you'll reach the web interface within the MultiVOIP GSM unit).

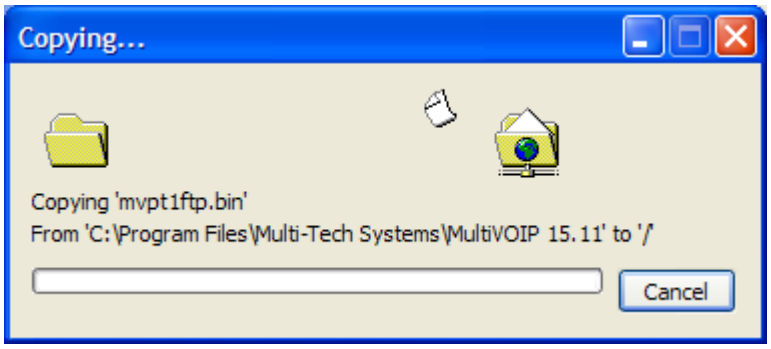

ftp://192.168.3.143/ - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Address ftp://192.168.3.143/
Other Places
Internet Explorer
My Documents
Shared Documents
My Network Places
Name Size Type Modified
casfile.cas 0 bytes CAS File 1/1/1980 12:00 PM
fdefftp.cnf 0 bytes SpeedDial 1/1/1980 12:00 PM
H323.pdl 0 bytes PDL File 1/1/1980 12:00 PM
mvpt1859ftp.bin 0 bytes BIN File 1/1/1980 12:00 PM
mvpt1ftp.bin 0 bytes BIN File 1/1/1980 12:00 PM
User: leonl
Internet

7. **Log In**. Use the User Name and password established in item #2 above. The login screens will differ depending on whether the FTP file transfer is to be done with a web browser or with an FTP client program (varies).

8. **Use Download**. Downloading can be done with a web browser or with an FTP client program.

Download with Web Browser:

- In the local Windows browser, locate the directory holding the MultiVOIP GSM program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP GSM *xxxx yyyy* (where x and y represent MultiVOIP model and software version numbers).
- Right-click on the file and select Copy files from Windows Explorer.
- Change focus to the Web browser window (left-clicking on the title bar will accomplish this), then right-click in the browser window showing the file and select Paste.
- You may be asked to confirm the overwriting of files on the MultiVOIP GSM. Do so.
- File transfer between PC and VOIP will look like a transfer within VOIP directories.



Download with FTP Client Program:

- In the local directory browser of the FTP client program, locate the directory holding the MultiVOIP GSM program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP GSM *xxxx yyyy* (where x and y represent MultiVOIP GSM model numbers and software version numbers).
- In the FTP client program window, drag-and-drop files from the local browser pane to the pane for the MultiVOIP GSM FTP server. FTP client interface operations vary. In some cases, you can choose between immediate and queued transfer. In some cases, there may be automated capabilities to transfer to multiple destinations with a single command.

9. **Verify Transfer**. The files transferred will appear in the directory of the MultiVOIP GSM. Check to insure that the new file is in place by verifying the file size and date (this is easy if this is the first time you have done this as it will change from a file size of '0' and a date of 01/01/1980).
10. **Log Out of FTP Session**. Whether the file transfer was done with a web browser or with an FTP client program, you *must* log out of the FTP session before opening the MultiVOIP GSM Windows interface.

Web Browser Interface

You can control the MultiVOIP GSM unit with a graphic user interface (interface) based on the common web browser platform. Qualifying browsers are Internet Explorer 6+, Netscape 6+, and Mozilla Firefox 1.0+.

MultiVOIP GSM Web Browser Interface Overview	
Function	Remote configuration and control of MultiVOIP GSM units.
Configuration Prerequisite	Local Windows interface must be used to assign IP address to MultiVOIP GSM.
Browser Version Requirement	Internet Explorer 6.0 or higher; or Netscape 6.0 or higher; or Mozilla Firefox 1.0 or higher.
Java Requirement	Java Runtime Environment version 1.4.0_01 or higher (available on your product CD)

The initial configuration step of assigning the VOIP unit an IP address must still be done locally using the Windows interface. However, all additional configurations can be done via the web interface.

The content and organization of the web interface is directly parallel to the Windows interface. For each screen in the Windows interface, there is a corresponding screen in the web interface. The fields on each screen are the same, as well.

The Windows interface gives access to commands via icons and pull-down menus whereas the web interface does not. The web interface, however, cannot perform logging in the same direct mode done in the Windows interface. However, when the web interface is used, logging can be done by email (SMTP).

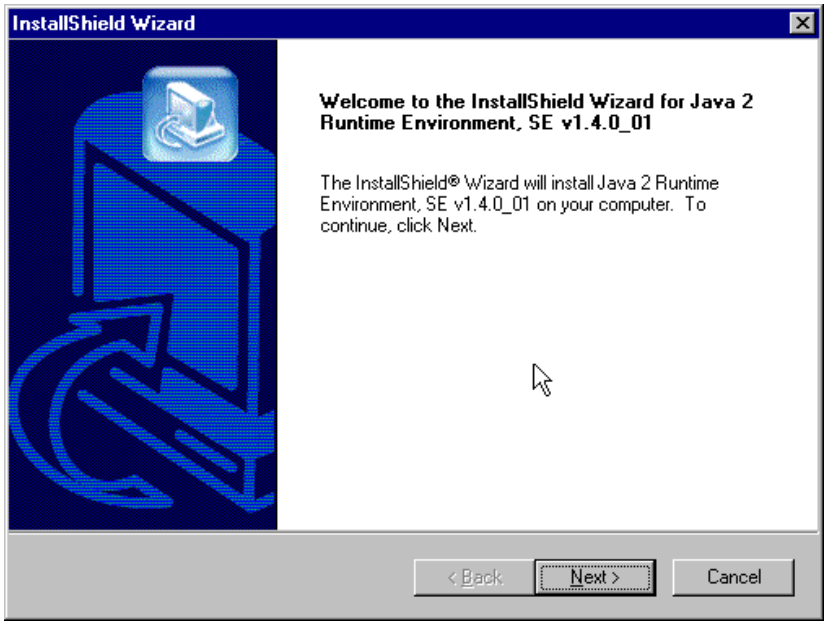
The graphic layout of the web interface is also somewhat larger-scale than that of the Windows interface. For that reason, it's helpful to use as large of a video monitor as possible.

The primary advantage of the web interface is remote access for control and configuration. The controller PC and the MultiVOIP GSM unit itself must both be connected to the same IP network and their IP addresses must be known.

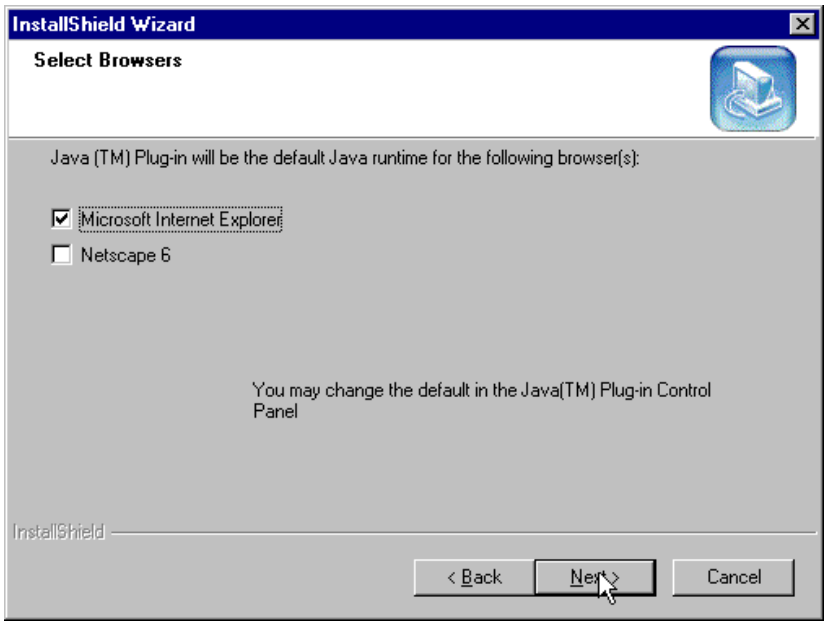
In order to use the web interface, you must also install a Java application program on the controller PC. This Java program is included on the MultiVOIP GSM product CD. Java is needed to support drop-down menus and multiple windows in the web interface.

Java Installation

To install the Java program, go to the **Java** directory on the MultiVOIP GSM product CD. Double-click on the .EXE file to begin the installation. Follow the instructions on the Install Shield screens.


InstallShield Wizard
Welcome to the InstallShield Wizard for Java 2 Runtime Environment, SE v1.4.0_01
The InstallShield® Wizard will install Java 2 Runtime Environment, SE v1.4.0_01 on your computer. To continue, click Next.
< Back
Next >
Cancel

During the installation, you may be asked to specify which browser(s) you'll use in the **Select Browsers** screen.


InstallShield Wizard
Select Browsers
Java (TM) Plug-in will be the default Java runtime for the following browser(s):
Microsoft Internet Explorer
Netscape 6
You may change the default in the Java(TM) Plug-in Control Panel
InstallShield
< Back
Next >
Cancel

When installation is complete, the Java program runs automatically in the background as a plug-in supporting the MultiVOIP GSM web interface. No user actions are required.

After the Java program has been installed, you can access the MultiVOIP GSM using the web browser interface. Close the MultiVOIP GSM Windows interface. Start the web browser. Enter the IP address of the MultiVOIP GSM unit. Enter a password when prompted. (A password is needed here only if password has been set for the local Windows interface or for the MultiVOIP GSM's FTP Server function. See "Setting a Password" earlier in this chapter.) The web browser interface offers essentially the same control over the VOIP as can be achieved using the Windows interface. As noted earlier, logging functions cannot be handled via the web interface. And, because network communications will be slower than direct communications over a serial PC cable, command execution will be somewhat slower over the web browser interface than with the Windows interface.

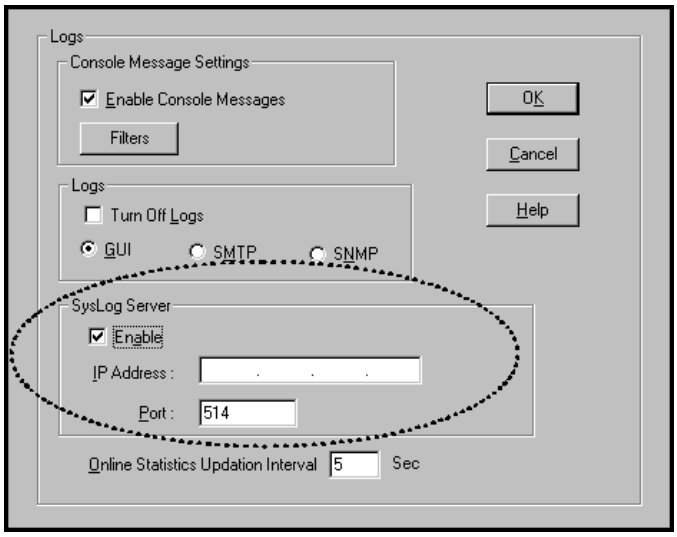
SysLog Server Functions

Multi-Tech has built SysLog server functionality into the software of the MultiVOIP GSM units. SysLog is a *de facto* standard for logging events in network communication systems.

To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon"). SysLog client programs, both paid and freeware can be obtained from Kiwi Enterprises (search the Internet for kiwi syslog daemon), among other firms. Read the End-User License Agreement carefully and observe license requirements. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

Multi-Tech Systems does not endorse any particular SysLog client program. SysLog client programs by qualified providers should suffice for use with MultiVOIP GSM units.

Before a SysLog client program is used, the SysLog functionality must be enabled within the MultiVOIP GSM in the **Logs** menu under **Configuration**.



Enable SysLog

The IP Address used will be that of the MultiVOIP GSM itself.

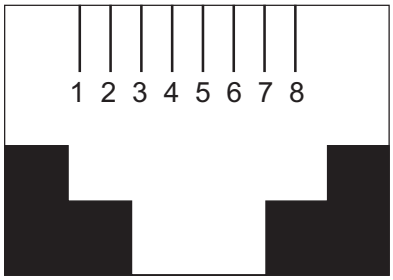
In the **Port** field, entered by default, is the standard ('well-known') logical port, 514.

Configuring the SysLog Client Program. Configure the SysLog client program for your own needs. In various SysLog client programs, you can define where log messages will be saved/archived, opt for interaction with an SNMP system, set the content and format of log messages, determine disk space allocation limits for log messages, and establish a hierarchy for the seriousness of messages (normal, alert, critical, emergency, etc.).

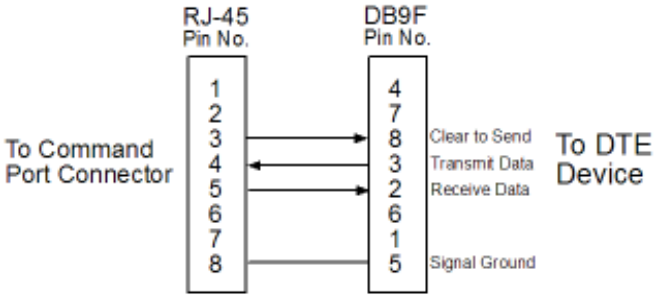
Appendix A – Ports & Cable Pin-outs

Command Cable

RJ-45 Connector



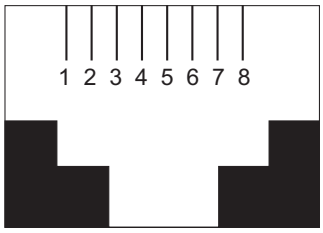
End-to-End Pin Info



RJ-45 connector plugs into Command Port of MultiVOIP GSM.
DB-9 connector plugs into serial port of command PC (which runs MultiVOIP GSM configuration software).

Ethernet Connector

The functions of the individual conductors of a MultiVOIP GSM Ethernet port are shown on a pin-by-pin basis below.

RJ-45 Ethernet Connector	Pin	Circuit Signal Name
	1	TD+ Data Transmit Positive
	2	TD- Data Transmit Negative
	3	RD+ Data Receive Positive
	4	-
	5	-
	6	RD- Data Receive Negative
	7	-
	8	-

Well Known Port Numbers

The following description of port number assignments for Internet Protocol (IP) communication is taken from the Internet Assigned Numbers Authority (IANA) web site (www.iana.org).

> "The Well Known Ports are assigned by the IANA and on most systems can only be used by system (or root) processes or by programs executed by privileged users. Ports are used in the TCP [RFC793] to name the ends of logical connections which carry long term conversations. For the purpose of providing services to unknown callers, a service contact port is defined. This list specifies the port used by the server process as its contact port. The contact port is sometimes called the "well-known port". To the extent possible, these same port assignments are used with the UDP [RFC768]. The range for assigned ports managed by the IANA is 0-1023."

Well-known port numbers especially pertinent to MultiVOIP GSM operation are listed below.

Port Number Assignment List:

Function	Port Number
telnet	23
tftp	69
http	80
ftp (simple)	115
SIP	5060
SMTP	25
SysLog	514

Appendix B – Regulatory Information

EMC, Safety, and R&TTE Directive Compliance

The CE mark is affixed to this product to confirm compliance with the following European Community Directives:
Council Directive 2004/108/EC of 31 December, 2004 on the approximation of the laws of Member States relating to electromagnetic compatibility,
and
Council Directive 2006/95/EC 12 December, 2006 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits,
and
Council Directive 1999/5/EC of 9 March 1999 on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity.

FCC Part 15 Class A Statement

This equipment has been tested and found to comply with the limits for a **Class A** digital device, pursuant to 47 CFR Part 15 regulations. The stated limits in this regulation are designed to provide reasonable protection against harmful interference in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy, and if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Plug the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the CFR 47 rules. Operation of this device is subject to the following conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference that may cause undesired operation.

Warning: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Industry Canada

This Class A digital apparatus meets all requirements of the Canadian Interference-Causing Equipment Regulations.

Cet appareil numérique de la classe A respecte toutes les exigences du Règlement Canadien sur le matériel brouilleur.

Canadian Limitations Notice

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

Appendix C – Additional Phone Book Examples

MVPGSM-to-MVPGSM

The basic setup would consist simply of two VOIPs able to communicate via IP address, with preset phone numbers entered into the Phone Book of each unit, with the Inbound Phone Book of one unit containing the exact same information as the Outbound Phone Book of the other unit and vice-versa. This is the underlying concept behind the Phone Books – they are the reverse of each other so that they match information to accomplish the connections. This is more applicable to analog VOIP units than the wireless, but it helps with understanding the purpose of the Phone Book sections.

Example 1: The *Local VOIP* has channel 1 populated with a SIM using the phone number 612-555-1234 and the *Remote VOIP* has 763-555-4321 as the phone number for the SIM on channel 1. By populating the Phone Book of each unit with the information from the other, calls can be changed as to appear local to the receiving unit. This is the most basic setup and illustrates the concept behind the Phone Book entries and standard VOIP operation. This direct connection then takes advantage of the wireless connection and the calls are not charged as long-distance.

Example 2: Multi-Tech VOIP units can also be set to Auto Call, in which case, either or both sides can be set to automatically dial the other VOIP ('612-555-1234' or '763-555-4321') when a phone number matches the entry set to Auto Call, the MVPGSM will automatically dial the corresponding phone number of the other VOIP unit.

MVPGSM as a Standalone Unit

Operating as a standalone unit (that is, with no corresponding VOIP unit on the receiving end), the MVPGSM can be used as a SIP gateway to handle all long distance calls and any other call that would be less expensive if the connection is wireless. In this case, the phone system call control module (e.g. an IP PBX) would determine the nature of the call being placed and route it to the proper gateway. Incoming calls should be routed using CLIP routing and/or a receptionist using the Auto Call feature.

Example 3: Your company has an IP PBX system with an analog VOIP unit for local calls and direct calls to the office in another city and an MVPGSM that will handle all long distance calls. With this setup, a call control module handles the call routing – when an extension requests an outside line (by dialing a "9" first, for example) the call control module then watches what the first digit after the 9 is. If that first digit is anything other than one of the local area codes, the call is routed through the MVPGSM to take advantage of the cost savings provided by the wireless network. If the first digit is a local call, it is sent through the standard PSTN. The analog VOIP could also be setup to connect with a remote office VOIP as well. In this case then, there would be another access digit (an "8" for example) to have an extension use the analog VOIP and connect to the remote location as if it were a local call.

Example Inbound Phone Books

The basic purpose of the Inbound Phone Book is to create rules for routing incoming IP calls. Below you will find the Inbound Phone Book settings for the examples that were detailed in the sample Phone Book Configurations above.

Example 1: Local VOIP has 763-555-4321 in the Inbound Phone Book while the Remote VOIP has 612-555-1234 in the Inbound Phone Book. A user at the Local VOIP can dial 612-555-1234 and it will ring the Remote VOIP, a user at the Remote VOIP can dial 763-555-4321 and it will ring the Local VOIP. Both calls are treated as local area code calls.

Example 2: Local VOIP has 763-555-4321 in the Inbound Phone Book while the Remote VOIP has 612-555-1234 in the Inbound Phone Book. A user at the Local analog VOIP can pick up an attached phone and it will ring the Remote GSM VOIP. The call is treated as a local area code call. *The Auto-Call feature is set in the Voice Parameters section*.

Example 3: The Inbound Phone Book for this setup will be set to Any Number for all channels, allowing all digits and relying on the call control module to handle the routing.

Example Outbound Phone Books

The Outbound Phone Book is the director of where calls will be routed – whether that is to a specific phone number (Any Number) or to the IP address of a remote VOIP unit for resending as a local call or out to the wireless network to take advantage of wireless plan savings. Below you will find the Outbound Phone Book settings for the examples that were detailed in the sample Phone Book Configurations above.

Example 1: Local VOIP has 612-555-1234 in the Outbound Phone Book while the Remote VOIP has 763-555-4321 in the Outbound Phone Book a. A user at the Local VOIP can pick up an attached phone and dial 763-555-4321 and it will ring the Remote VOIP, a user at the Remote VOIP can pick up an attached line and dial 612-555-1234 and it will ring the Local VOIP. Both calls are treated as local area code calls.

Example 2: Local VOIP has 612-555-1234 in the Outbound Phone Book while the Remote VOIP has 763-555-4321 in the Outbound Phone Book a. A user at the Local VOIP can pick up an attached phone and it will ring the Remote VOIP, a user at the Remote VOIP can pick up an attached line and it will ring the Local VOIP. Both calls are treated as local area code calls.

The Auto-Call feature is set in the Voice Parameters section.

Example 3: The Outbound Phone Book is set to the destination pattern number of the auto attendant. In this example the MVPGSM would also be setup to use a proxy (the call control module).

Appendix D – Waste Electrical and Electronic Equipment (WEEE) Statement

July, 2005

The WEEE directive places an obligation on EU-based manufacturers, distributors, retailers and importers to take-back electronics products at the end of their useful life. A sister Directive, ROHS (Restriction of Hazardous Substances) complements the WEEE Directive by banning the presence of specific hazardous substances in the products at the design phase. The WEEE Directive covers all Multi-Tech products imported into the EU as of August 13, 2005. EU-based manufacturers, distributors, retailers and importers are obliged to finance the costs of recovery from municipal collection points, reuse, and recycling of specified percentages per the WEEE requirements.

Instructions for Disposal of WEEE by Users in the European Union

The symbol shown below is on the product or on its packaging, which indicates that this product must not be disposed of with other waste. Instead, it is the user's responsibility to dispose of their waste equipment by handing it over to a designated collection point for the recycling of waste electrical and electronic equipment. The separate collection and recycling of your waste equipment at the time of disposal will help to conserve natural resources and ensure that it is recycled in a manner that protects human health and the environment. For more information about where you can drop off your waste equipment for recycling, please contact your local city office, your household waste disposal service or where you purchased the product.

Appendix E – C-ROHS HT/TS Substance Concentration

依照中国标准的有毒有害物质信息

根据中华人民共和国信息产业部 (MII) 制定的电子信息产品 (EIP)
标准－中华人民共和国《电子信息产品污染控制管理办法》（第 39 号），也称作中国
RoHS，下表列出了 Multi-Tech Systems Inc. 产品中可能含有的有毒物质 (TS) 或有害物质 (HS)
的名称及含量水平方面的信息。

成分名称	有害/有毒物质/元素					
	铅 (PB)	汞 (Hg)	镉 (CD)	六价铬 (CR6+)	多溴联苯 (PBB)	多溴二苯醚 (PBDE)
印刷电路板	O	O	O	O	O	O
电阻器	X	O	O	O	O	O
电容器	X	O	O	O	O	O
铁氧体磁环	O	O	O	O	O	O
继电器/光学部件	O	O	O	O	O	O
IC	O	O	O	O	O	O
二极管/晶体管	O	O	O	O	O	O
振荡器和晶振	X	O	O	O	O	O
调节器	O	O	O	O	O	O
电压传感器	O	O	O	O	O	O
变压器	O	O	O	O	O	O
扬声器	O	O	O	O	O	O
连接器	O	O	O	O	O	O
LED	O	O	O	O	O	O
螺丝、螺母以及其它五金件	X	O	O	O	O	O
交流-直流电源	O	O	O	O	O	O
软件/文档 CD	O	O	O	O	O	O
手册和纸页	O	O	O	O	O	O
底盘	O	O	O	O	O	O

X 表示所有使用类似材料的设备中有害/有毒物质的含量水平高于 SJ/Txxx-2006 限量要求。

O 表示不含该物质或者该物质的含量水平在上述限量要求之内。

INDEX

A

Antenna Requirements....8
Auto Call Example....79
Auto Disconnect....39
AutoCall....39

B

Basic Setup....17

C

Cabling....11
Call Forwarding Example....81
Call Progress fields....95
Call Routing....63
Calling Plan....43
Change Wireless Band....12
CLIP Routing....63
CLIP Routing Example....78
Coder Parameters fields....38
Comm. Port Unavailable....68
Configuration Examples....70
Creating a User Default Configuration....67
Custom Tones and Cadences....52

D

Default IP Address....32
Diff Serv PHB value....35
DTMF inband....37
DTMF out of band....37
Dynamic Jitter....39

E

Email log reports....53
Endpoint Statistics....98

F

FTP Server function....111
FTP Server, logging out....114

I

Inbound Phone Book Example....75
IP Statistics....101
IP Statistics fields....102

L

LED descriptions....6
Link Management fields....103
Load Balance
  First Available Hunting....41
  Least Used First....41
  Round Robin....41
Load Balancing....41
Load Balancing Example....72
Local Survivability....65
Local Survivability Examples....82
Logs....99
Logs (Statistics) field definitions....100

M

MultiVOIP GSM Not Found....68

N

NAT Traversal screen fields....60

O

Outbound Phone Book Example....77

P

Packet Prioritization 802.1p....34
Packetization rates....105
Password Recovery....109
Password Reset....109
Phone Book....69
Phone Database not Read....68

R

RADIUS....56
RADIUS Screen field definitions....57
Regional parameter definitions....51
Regional Parameters....50
Regulatory Information....120

S

Saving the MultiVOIP GSM Configuration....67
Set Baud Rate....67
Set Log Reporting Method....58
Setting Ethernet/IP parameters....33
Setting password....108
Setting user defaults....107

SIP Call Signaling 45
SIP Call Signaling parameter definitions 45
SIP Server Definitions 49
SMTP 53
SMTP parameters definitions 54
Software Installation 14
Specifications 7
STUN clients and servers 60
Supplementary Services 62
SysLog 118

T

Technical Support 2

V

Voice/FAX parameter definitions 36

W

Warranty 2
WEEE Statement 124
Wireless Interface parameters 40
Wireless Statistics 97